DYNAMIX DW-0101/H
DYNAMIX DW-0202/H

FXSO Gateway

User Manual

Steps in configuration

STEP 1 **Start Up**	**To check out the peripheral equipments and understand the feature of this gateway. Please read this step very carefully before starting the configuring.**
STEP 2 **How to Setup and Connect basically**	**Connecting the gateway and computer to start configuring by WEB GUI.** **Setting the ip address for this gateway to make sure that it could connect with the internet.** **Setting the configurations of dialing, including the Peer-To-Peer, GK mode and how to set these tables to make calls by this gateway easily.** **The other configurations of make call will be discussed in this step.**
STEP 3 **Advanced**	**Advanced configurations and special functions of this gateway. Using the WEB GUI to show how to set this table and explain the meaning of these tables.**
STEP 4 **Command List**	**To explain the meaning of the command in the command line interface and example the usage of the command.** **To get more usages or configuration in this step and study about the command line configuration.**

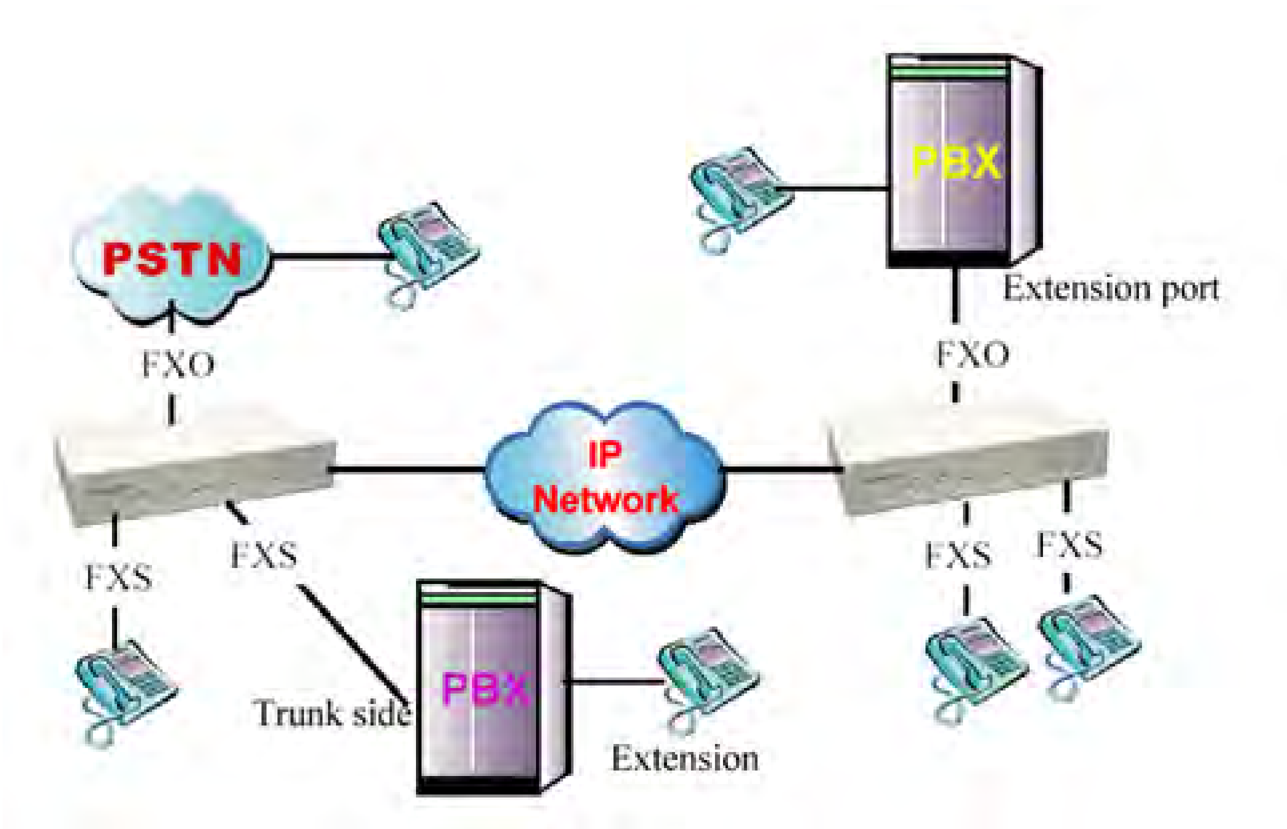
1. Start Up

1.1 Introduction

Dynamix DW-0101/H / Dynamix DW-0202/H is a one-port/two-port FXS+FXO gateway. It supports an innovative intelligent call rouging function that transparently routes calls to destination either through PSTN or Internet.

Dynamix DW-0101/H / Dynamix DW-0202/H provides voice over IP and FAX over IP services for ITSP/ISP Internet Telephony Services Provider and Office/SOHO IP-PBX application.

Application Architecture



FXO ports can connect with PSTN Line or Extension Line of PBX
FXS ports can connect with Phone Set or Trunk Line of PBX

1.2 Features and specification

Features

- ITU-T H.323 v2/v3/v4 compliance
- Automatically Dial Path Selection (IP or PSTN)
- PSTN Line switch to telephone set when power is failure
- PPPoE support
- Behind NAT router or IP sharing device
- DNS server inquiry
- Automatically Gatekeeper Discovery
- Provide Peer-to-Peer Mode (Non Gatekeeper needed) selection
- E.164 Dial Plan
- TFTP/FTP software upgrade
- Remote configuration/ reset
- LED indication for system status
- MS-NetMeeting v3.0 compatible
- Support Fix IP and DHCP
- Parameters of VoIP packet

Audio feature

- Codec -- G.711 a/µlaw, G.723.1 (6.3kbps), G.729, G.729A
- G.168/165-compliant adaptive echo cancellation
- Dynamic Jitter Buffer
- Completed voice band signaling support
- Provide In-band or Out-band DTMF generation/detection
- Provide call progress tone

Management Feature

- TELNET/Console port and Web Browser configuration

Certification

- UL, CE, FCC

FXS Features

- 2-wire loop start
- Programmable On-Hook voltage, Ring voltage/Cadence/Frequency, Loop current
- Line polarity reversal generation

FXO Features

- 2-wire loop start
- Support auto-attendant (Tone or voice greeting)
- PSTN polarity reversal detection
- Provide 2^{nd} dial tone to PSTN
- Disconnect tone detection
- Asking ping function with the inside or outside calls

Environmental

- Operation temp:0°C to 40°C
- Humidity: 10% to 90% (Non-condensing)

1.3 Accessories and equipment

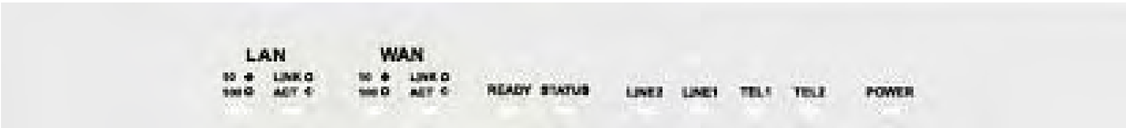
- The voice gateway in 2/4 FXS ports models and two RJ-45 connector (WAN and LAN).
- The AC adapter.
- The CD of user manual.
- The connection cable in RS-232 interface.

1.4 Appearance

Front panel: The LED lights provide related system messages of the gateway.

Dynamix DW-0101/H

Dynamix DW-0202/H



Power: Light on means Gateway is power on, and vice versa.
TEL: Light on means the line is in use (off-hook), and vice versa.
LINE: Light on means the line is in use (off-hook), and vice versa.
Status:

1. LED light on means Gateway has successfully registered to Gatekeeper when it is in Gatekeeper Mode.
2. LED flash means Gateway is not registered to Gatekeeper when it is in Gatekeeper Mode.
3. Or when Gateway is in downloading mode, LED should be flash as well.
4. LED light off means Gateway is in Peer-to-Peer Mode.

Ready:

1. Light on and slow flash means Gateway is in normal mode.
2. Light on and fast flash means Gateway is in downloading mode.

WAN: Connected to Public Ethernet

1. Line- LED light on means Gateway is physically connected to the Ethernet correctly.
2. ACT- LED light on and flash when Ethernet data is being transmitted / received.

LAN: Switch to another device, such as PC

1. Line- LED light on means Gateway is physically connected to the Ethernet correctly.

2. ACT- LED light on and flash when Ethernet data is being transmitted received.

Back panel:

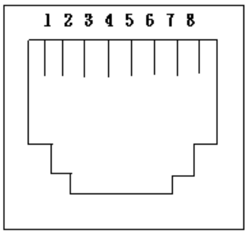
Dynamix DW-0101/H

Dynamix DW-0202/H

1. **Ethernet Port**

   LAN/WAN: 10/100 Base-T; RJ-45 socket, complied with ETHERNET 10/100base-T.

   The pin-out is as following:



PIN 1, 2: Transmit

PIN 3, 6: Receive

2. **COM:**

   RS232 console port (DB-9pin **male** connector)

   Note: use straightforward cable to connect to your computer.

PINOUTS

Pin	Name	Dir	Description
2	RXD	←	Receive Data
3	TXD	→	Transmit Data
5	GND	—	System Ground

3. **TEL:**
   RJ-11 connector, FXS interface is for connecting the analog phone sets or trunk line of PABX.
4. **LINE:**
   RJ-11 connector, FXO interface is for connecting the extension line of PABX or PSTN Line.
5. **12V DC:**
   Input AC 100V~120V;output DC12V.

2. How to Setup and Connect basically

2.1 System Requirement

1. One PC (a) Pentium 100 or above, 64 RAM, Windows 98 or above.
   (b) Ethernet card or COM port
2. One standard straightforward RS-232 cable (female connector to Gateway side).
3. Analog telephone sets.
4. PBX extension Lines or PSTN Lines.
5. Software tools (a) Hyper Terminal, TELNET, Web Browser.
   (b) Gatekeeper (optional).

2.2 IP Environment Setting

User must prepare a valid IP address, complied with IP Network, for Gateway's proper operation.

For testing the validation of chosen IP address, using the same IP configuration in other PC or Notebook, and then try to connect to Public Internet (go to well-known website, receive Internet mail, or ping a specific public IP address). If it works, use the same IP address and network configuration for Gateway. Please follow up the step for the configuration of your computer or notebook.

2.2.1 For Windows 2000/NT

Please make sure that the network interface of your computer is working fine and the cross over line (RJ-45) is connecting with the computer correctly or you could use a hub to connect with your computer and this gateway. Turn on your computer and configure the network parameter as follow:

1 Go to the **start** menu and enter the **setting** area. Click **control panel**.

2 Enter the network configuration.

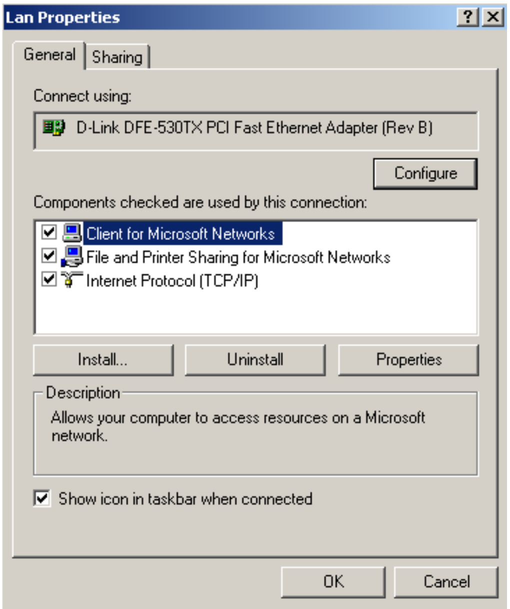


Figure 2.1: Network Configuration

3 Select the **Property** of the LAN card.

4 Setup the ip address, subnet mask and default gateway as below:

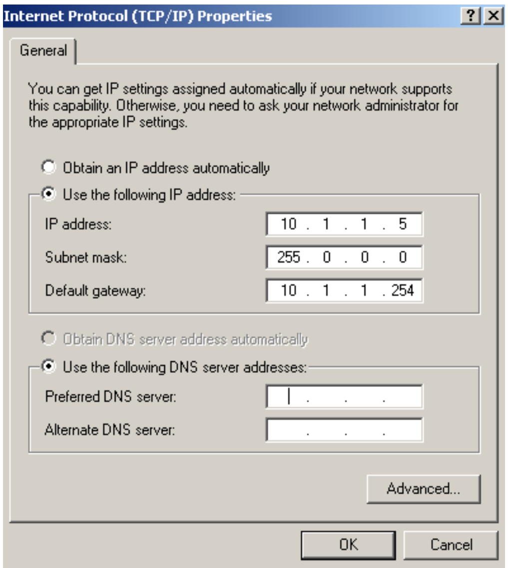


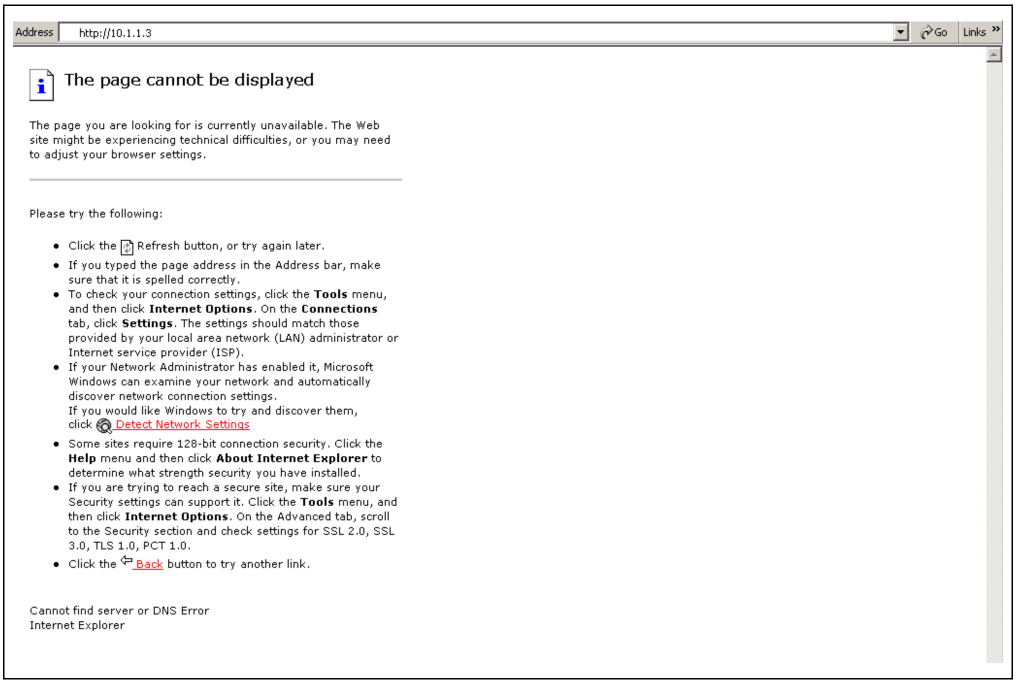
Figure 2.2: Configure the network

5 Click OK after you finished the network setup.

The default ip address, netmask and default gateway address of the gateway is 10.1.1.3, 255.255.0.0, 10.1.1.254.

2.3 Network configurations in your gateway

1 Key in the ip address of the gateway (http://10.1.1.3) with the browser



2 After key in the ip address, you have to enter the user name and password to enter the WEB configuration. (Username: root ; No password)

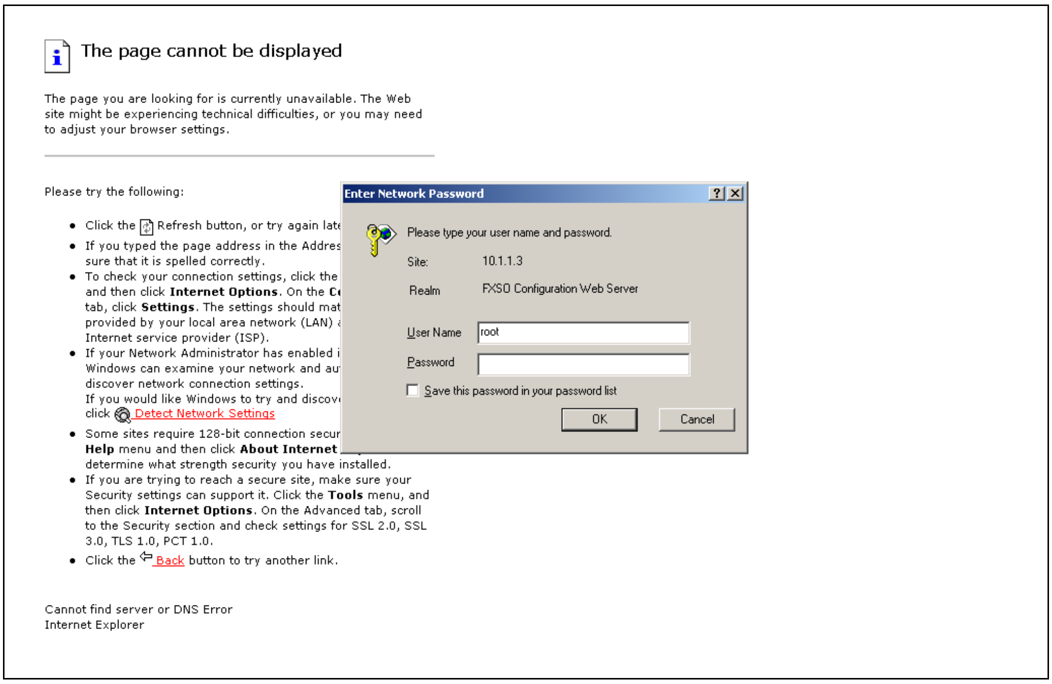


Figure 2.3: Login the username and password

3 You will enter the main page of the configuration after key in the login name and password correctly:

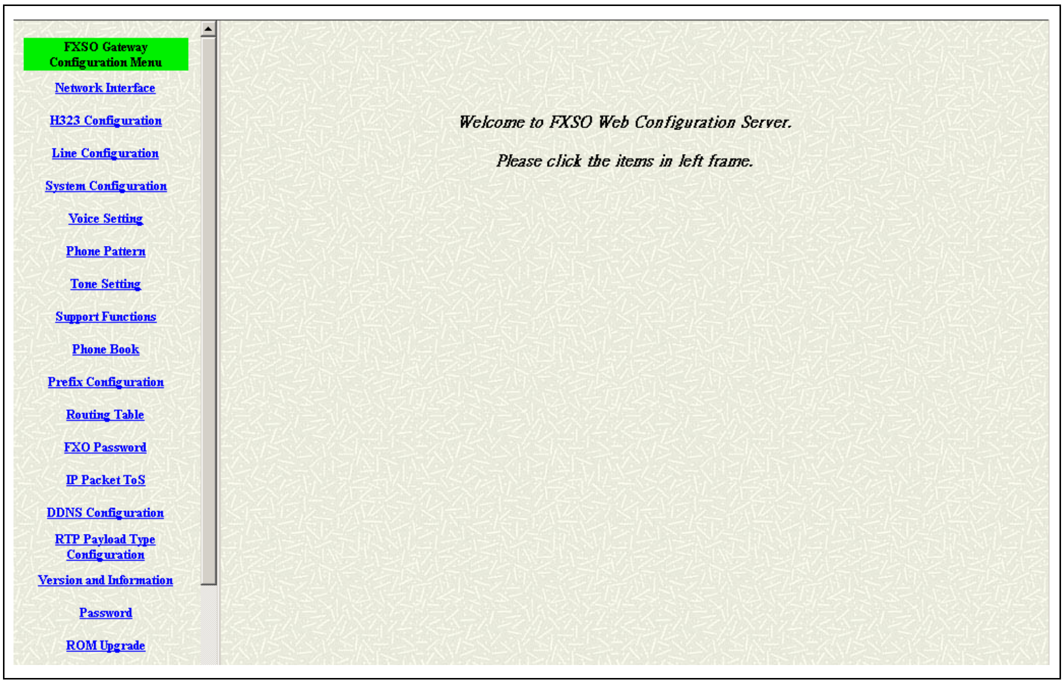


Figure 2.4: The main WEB configuration

4 Press the **Network Interface** to configure the networking of your gateway:

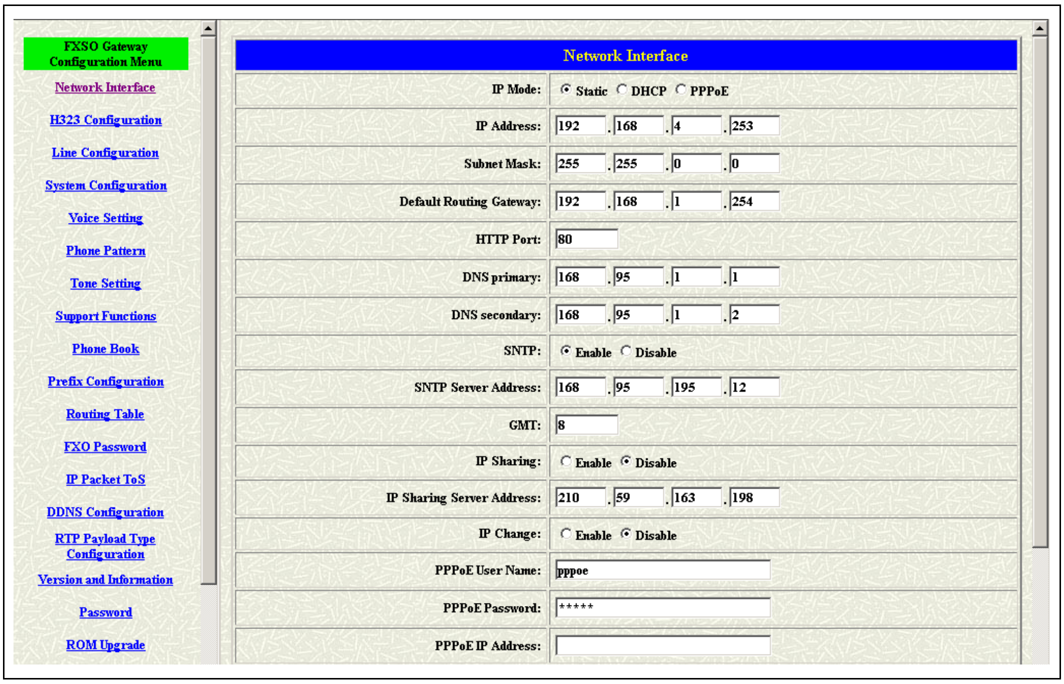


Figure 2.5: The Network Interface

2.3.1 Static ip address

1 Please get the correct ip address, netmask and default gateway address from your ISP first. Press the OK button if you finished.

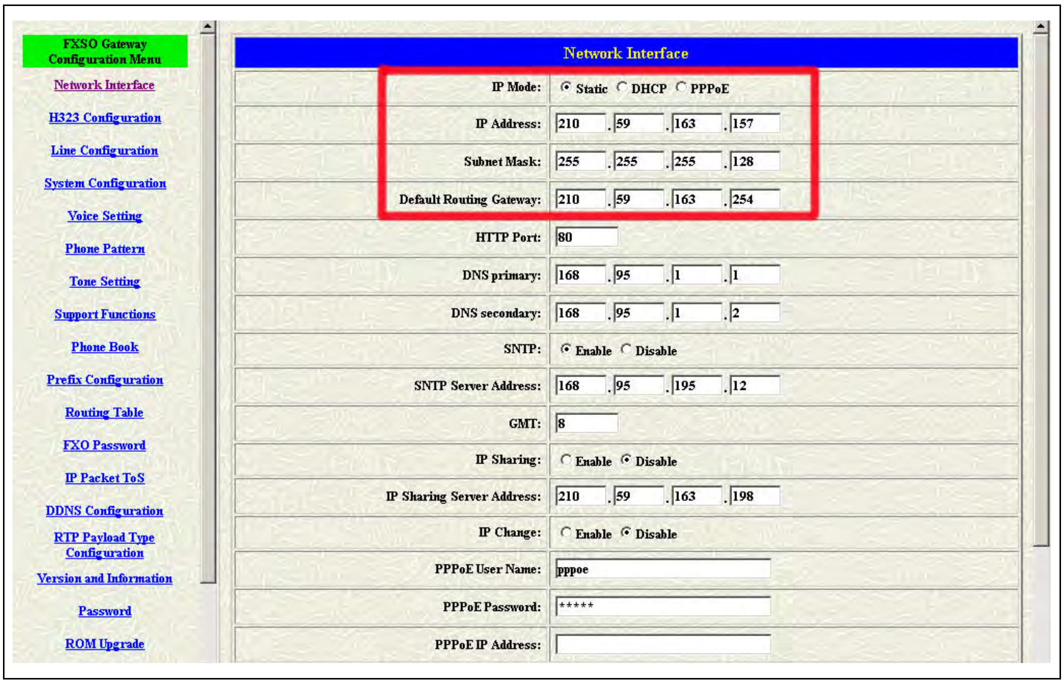


Figure 2.5: Configure the static ip address

2 Press the commit if you finish the configuration.

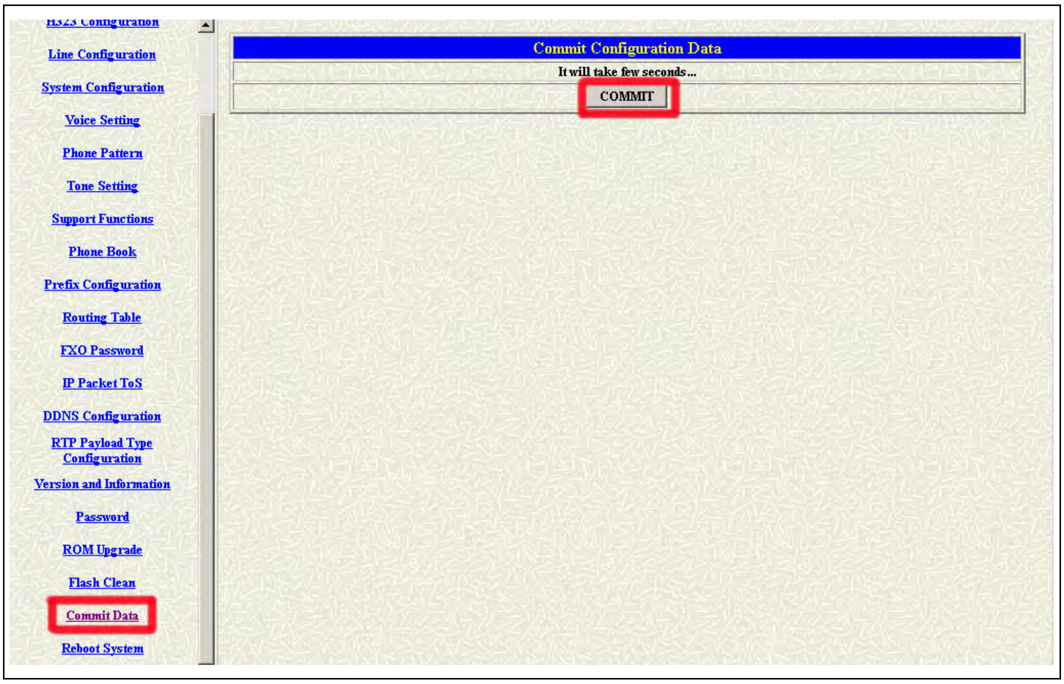


Figure 2.6: Commit the data

3 Press the reboot if you want the configuration executed.

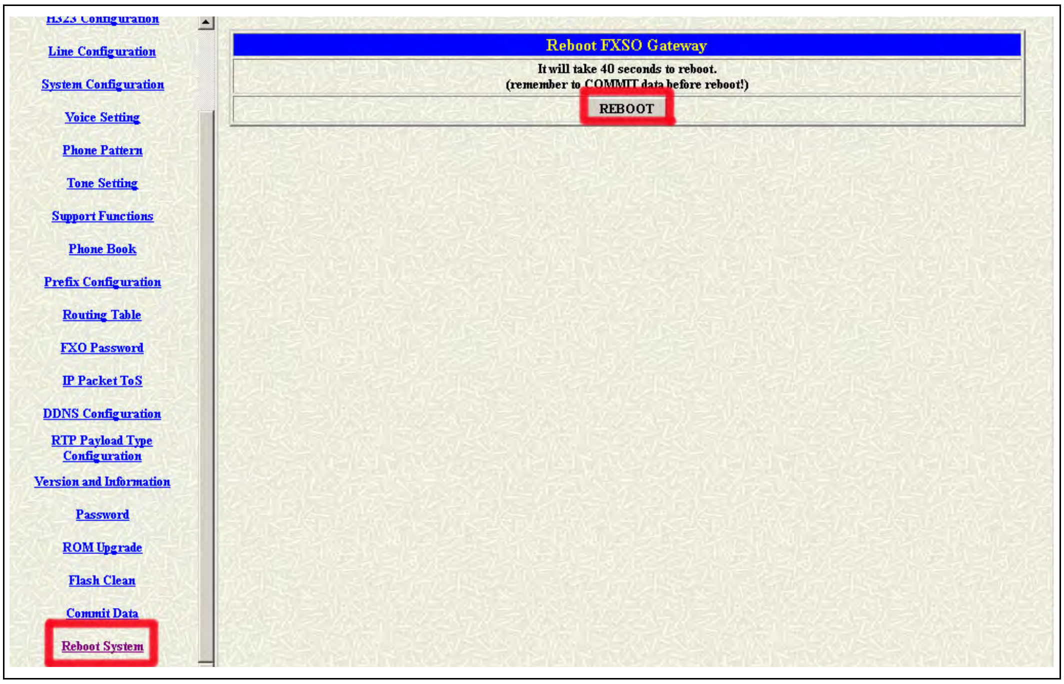


Figure 2.7: Reboot the system

2.3.2 DHCP mode

1 Change to the DHCP mode if you are using the cable modem or DHCP server.

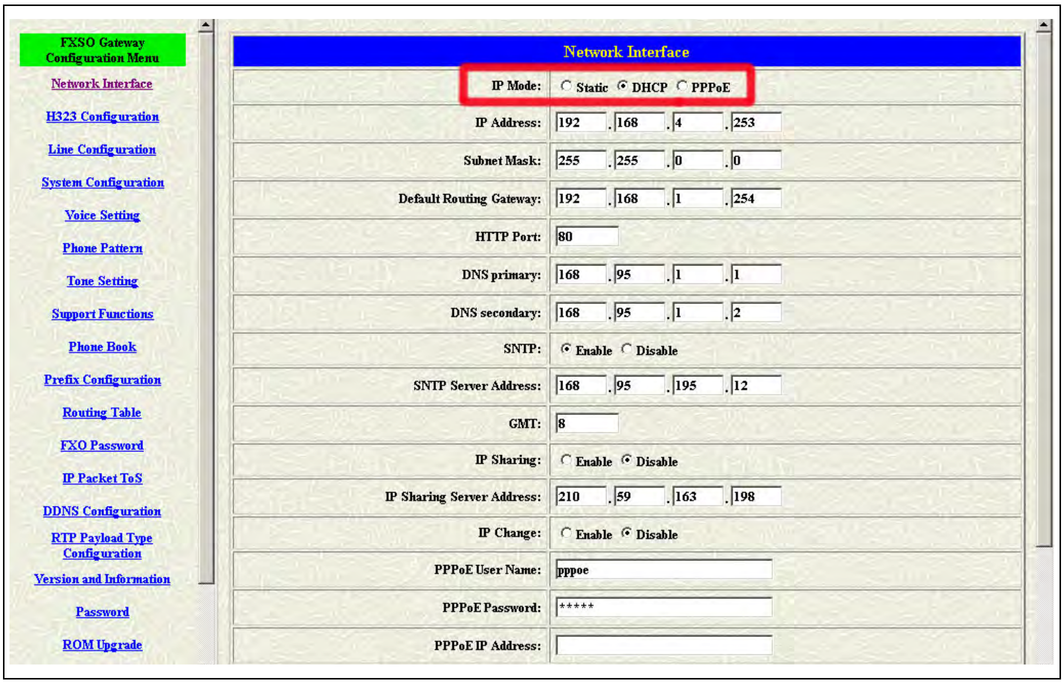


Figure 2.8: Enable the DHCP function

2 Please commit the data and reboot the machine after you enable the DHCP function.

2.3.3 PPPoE mode

1 Chang the PPPoE mode in the IP mode table.

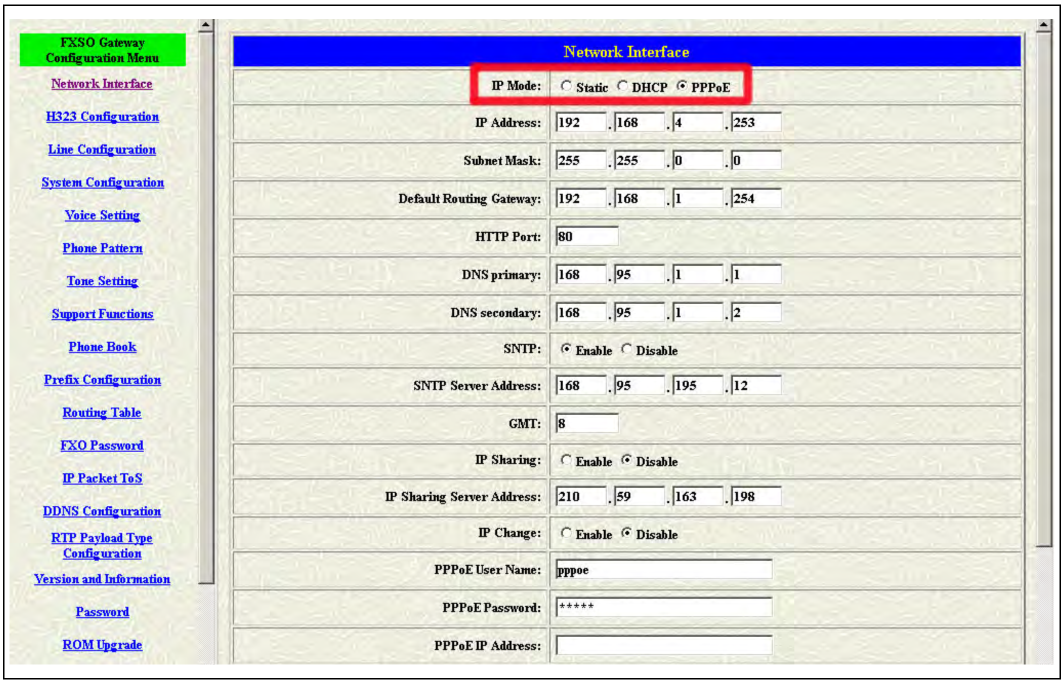


Figure 2.9: Chage to PPPoE mode

2 Please commit the data and reboot the machine after you finished the configuration of PPPoE.

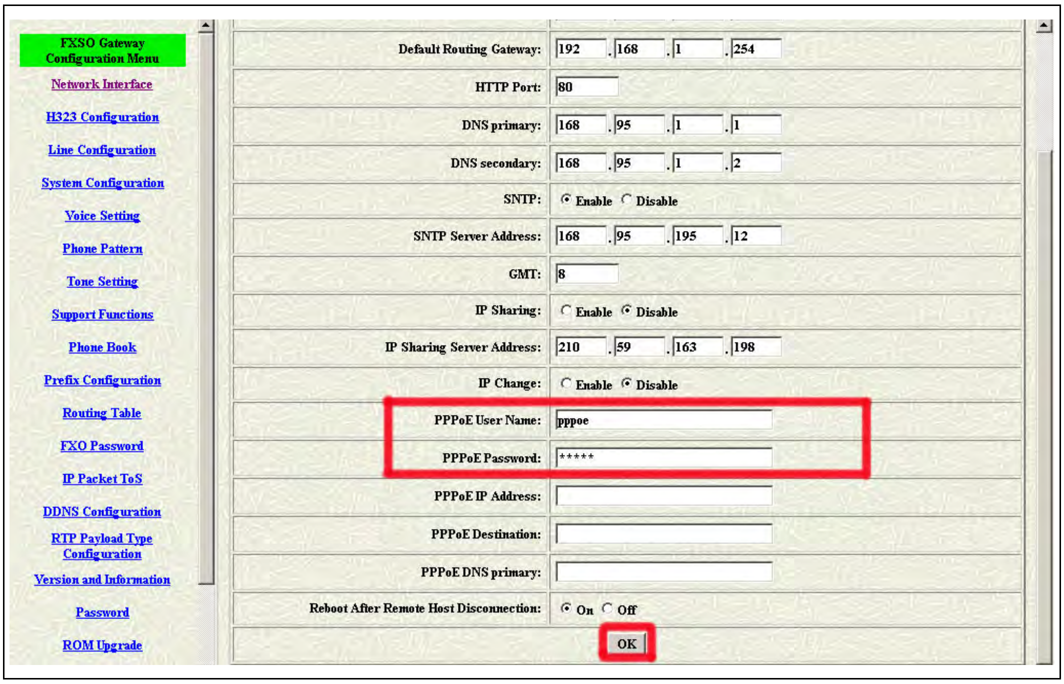


Figure 2.10: Enter the PPPoE connection info

2.4 Making a VoIP Call

There are two modes that you could configure the gateway for making VoIP calls. One is the Peer-to-Peer mode, another is GK routed mode. The configurations and functions are different. Please make sure about the mode you want and follow up the step to configure your gateway.

2.4.1 Configure the gateway into the Peer-to-Peer mode

1 Enter the H323 Configuration table and change the mode to Peer-to-Peer.

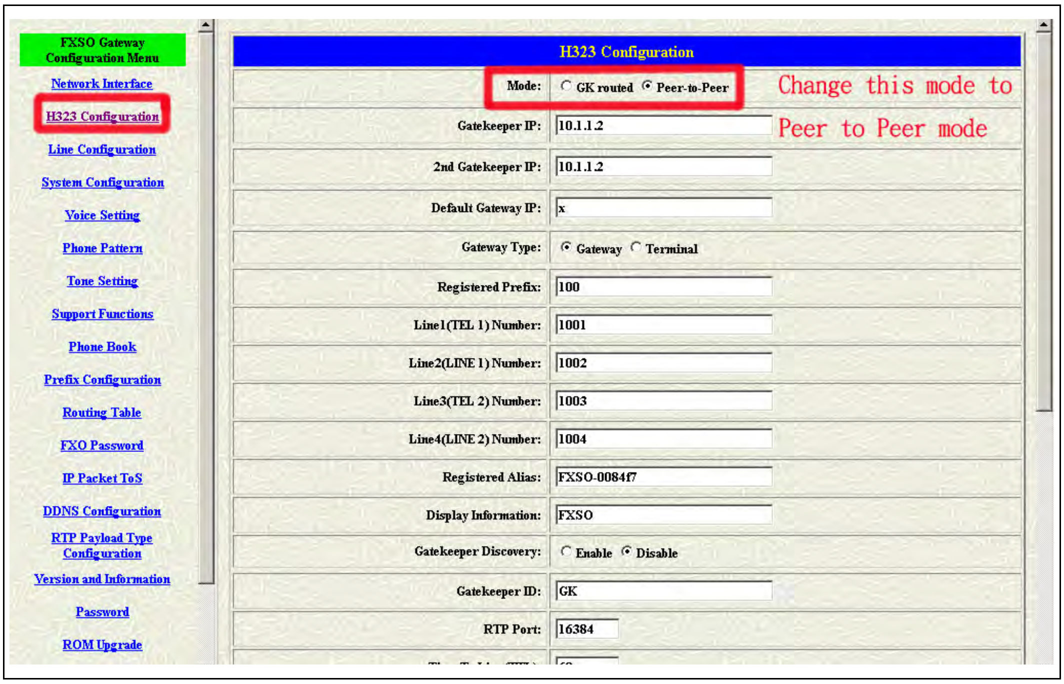


Figure 2.10: Configure the Peer-to-Peer mode

2 Press the OK button which is on the buttom of this page to save the configuration.

Figure 2.11: Press OK to save the data

3 Enter the Phone Book configuration table and configure the name, ip address and phone number of the destination.

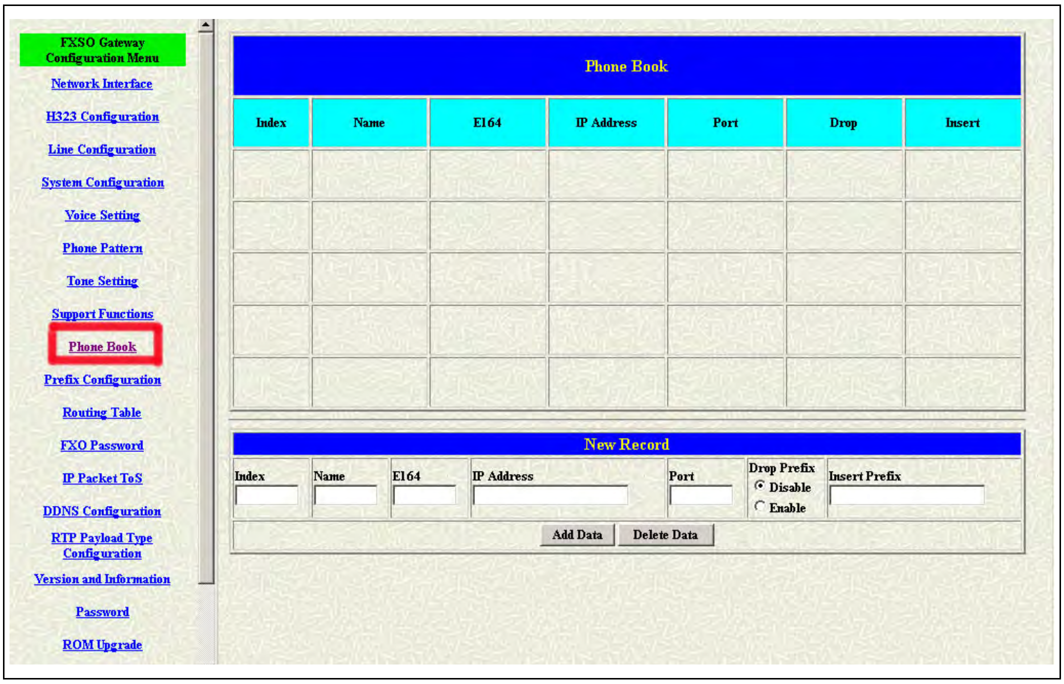


Figure 2.12: Phone Book

Example

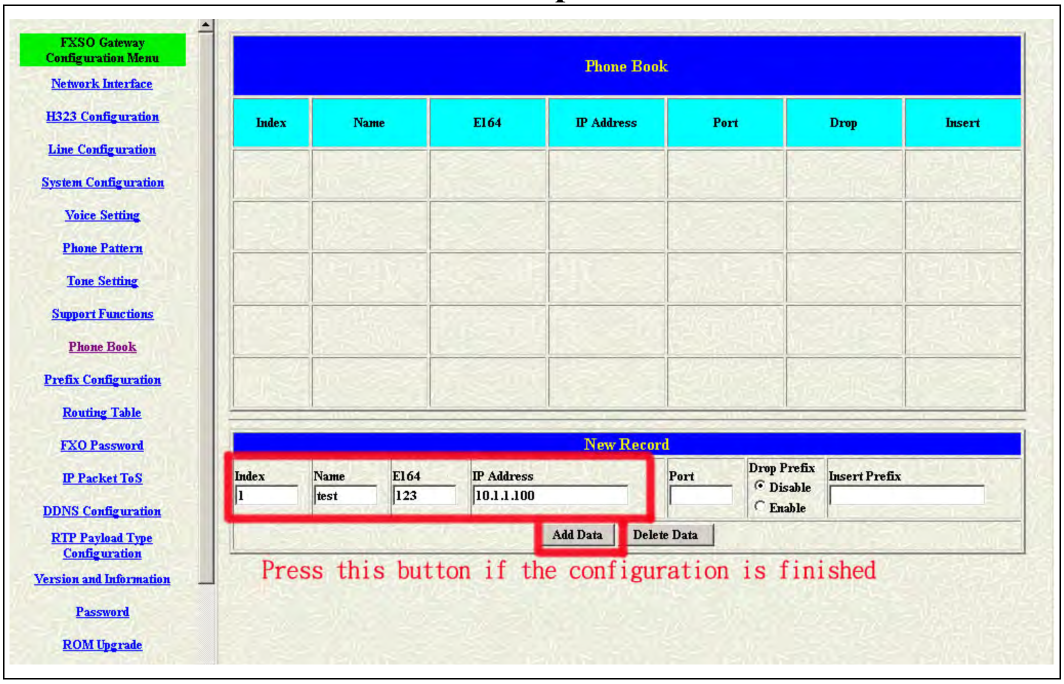


Figure 2.13: The example of Phone Book configuration

The name of the destination: **test**
The E164 number (phone number) of the destination: **123**
The ip address of the destination: **10.1.1.100**
The call signal port of the destination: **1720**
(The port will be 1720 if you don' t define it)
Drop prefix: **Enable – The e164 number you define will be deleted**
Disable – The e164 number you define will be kept
Insert prefix: **To add a number you define in this table**

Press the "Add Data" button when you finished, and the new table will display on the first index if you press the Phone Book configuration button.

4 Please Commit it and Reboot the system if the configuration is finished.

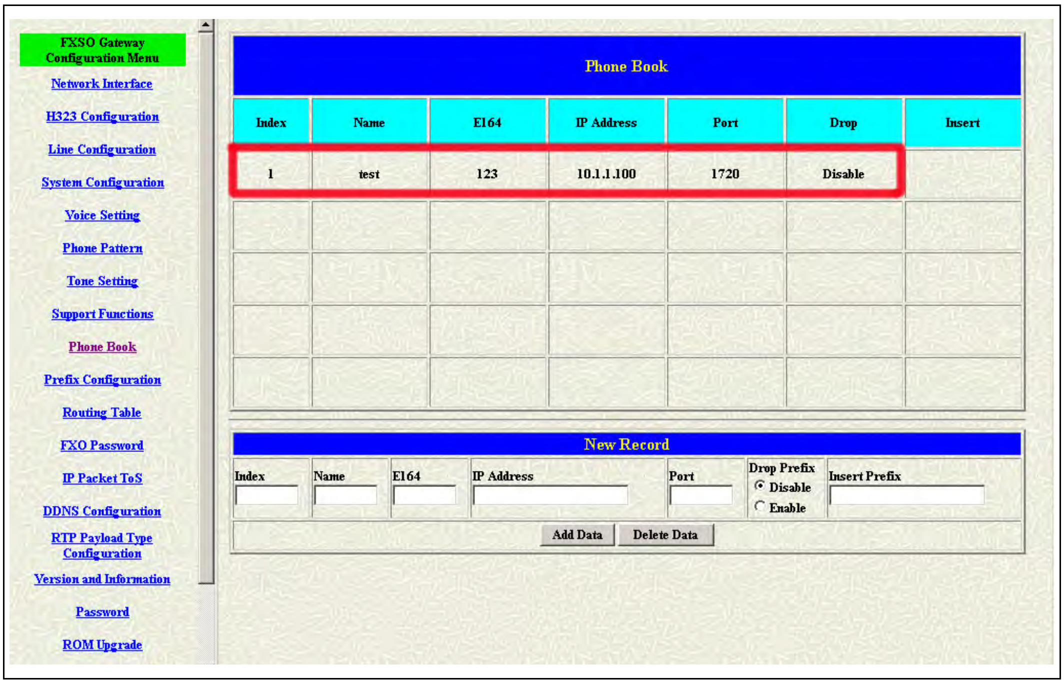


Figure 2.14: To show the Phone Book record

Phone Book is only for the Peer-to-Peer mode.
Fifty index support.

The application in the drop and insert function

Input (E164)	Drop	Insert	Output
100	Disable	X	100
200	Disable	0	0200
300	Enable	X	X
400	Enable	500	500

X – Do not enter any numbers

2.4.2 Configure the gateway into the GK routed mode

1 Enter the H323 Configuration table and change the mode from Peer-to-Peer to GK routed. To change the GK information from your service provider (Ex: The Gatekeeper IP, Registered Prefix and Registered Alias).

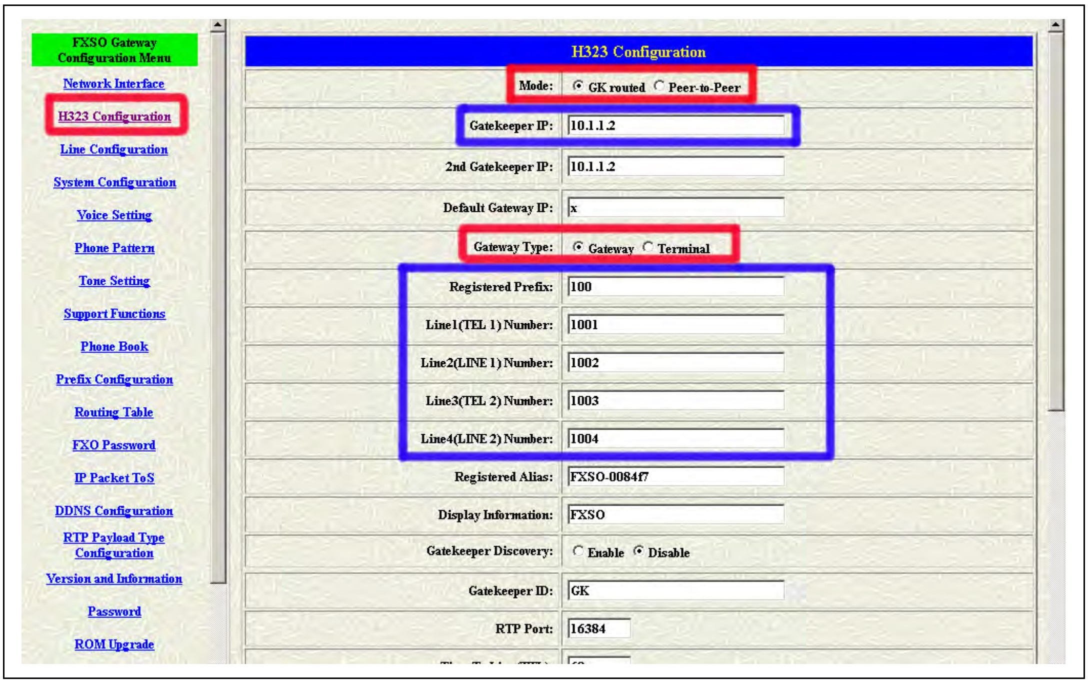


Figure 2.15: Configure the GK info

2 Press the OK button which is on the buttom of this page to save the configuration.

Figure 2.16: Press OK to save the data

3 Press the Commit Data and Reboot System buttons when you finished the configuration.

2.4.2.1 The type in GK routed mode

There are two types in the GK routed mode you could choose. One is Gateway type and another is Terminal type. There are some different functions, applications and configurations between the Gateway type and Terminal type. Please get more info from the table as below:

Gateway Type	Terminal Type
Prefix number is necessary	**Prefix number is not necessary**
The E164 number of each port should be follow the prefix number	**The E164 number of each port could be different**
Support the one-stage dialing	**Do not support the one-stage dialing**

The default type of the all series Asotel gateway are Gateway type. But the LanPhone is the terminal type. You could change the type for your gateway but the LanPhone couldn't.

Please pick up the type you want to use with your gateway and pay more attentions about the rules as above.

There are two example for the Gateway type and Terminal type as below:

Example – Gateway type

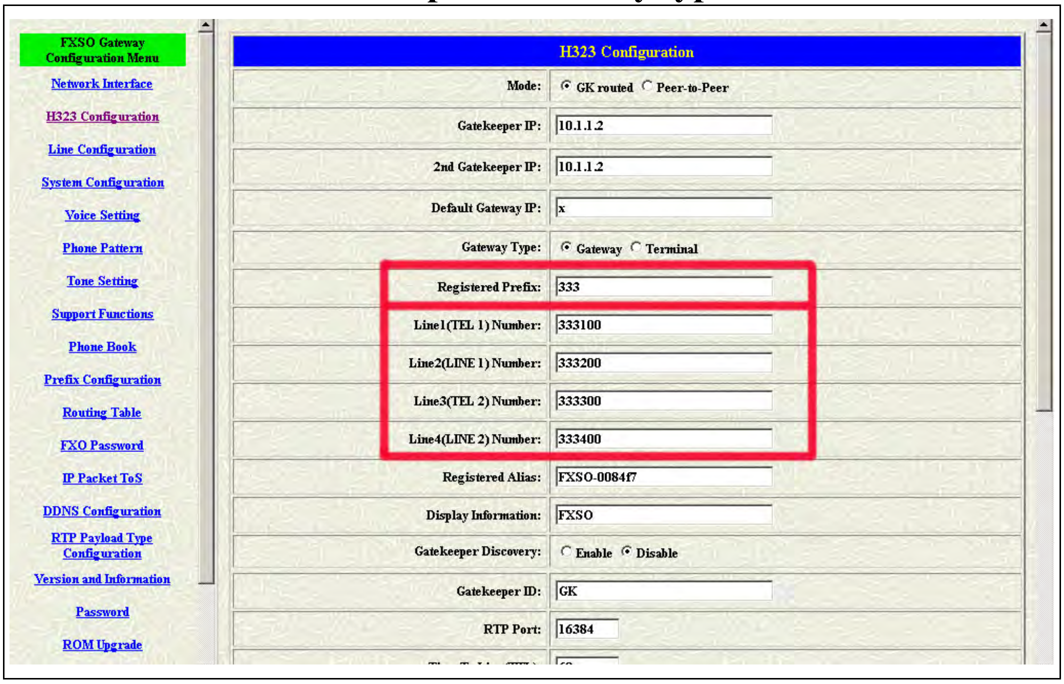


Figure 2.17: Gateway Type configuration

The prefix number is necessary if you configure it to be a gateway. The number of ports must be followed the prefix number.

Example – Terminal type

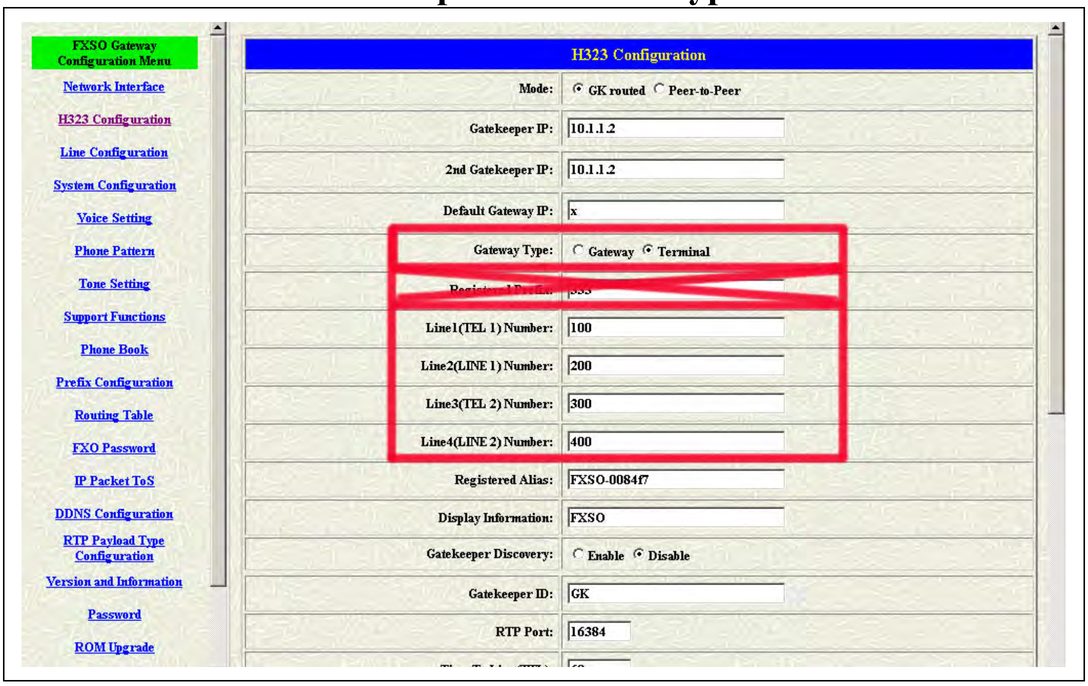


Figure 2.18: Terminal Type configuration

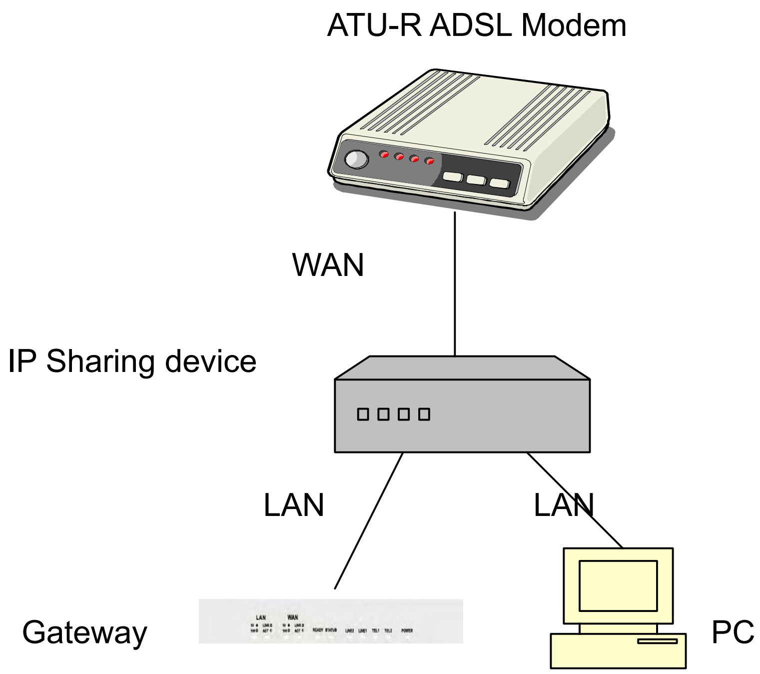
The prefix number is only for the gateway type and the number of ports could be configured in the different numbers.

3. Advanced

There are too many advanced commands for the advanced users. The following chapters are based on the application layer. Please get the info what you need. If you need the command, please watching the chapter of Command Line Interface.

3.1 IP Sharing

The function is only for the user who is using the IP Sharing device. It is said Gateway is connected to the IP Sharing device.
The IP Sharing Device must support the DMZ or Virtual server functions
An example such as ADSL network is in the following.



- The WAN IP Address obtained from ADSL has two kinds of methods.
  One is fixed IP Address, while user applies for one or more fixed IP Addresses.
  Another is dynamic IP Address while user applies for dial-up connection way.

◆ The LAN IP Address of User's PC can be set as DHCP client in order to gain a valid one.

◆ Another IP Address for Gateway must be set as an fixed one in order for that IP Sharing device pass forwarding the relevant information from WAN to LAN. Besides, a valid IP Address meets the IP Sharing device (LAN site) is the element.

◆ Please configure your gateway just like below:

1 Enter the IP address, Netmask and the default gateway in the network table. Please follow up your IP Sharing device.

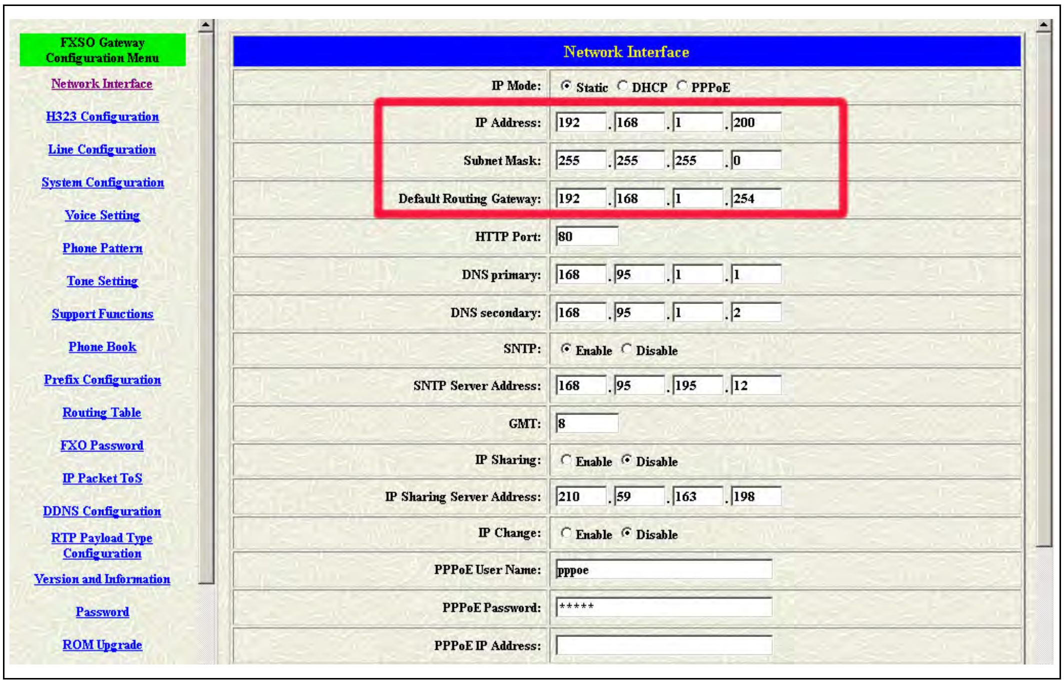


Figure 3.1: Network configuration

2 Enable the ip sharing function and put the static ip address in the server address table.

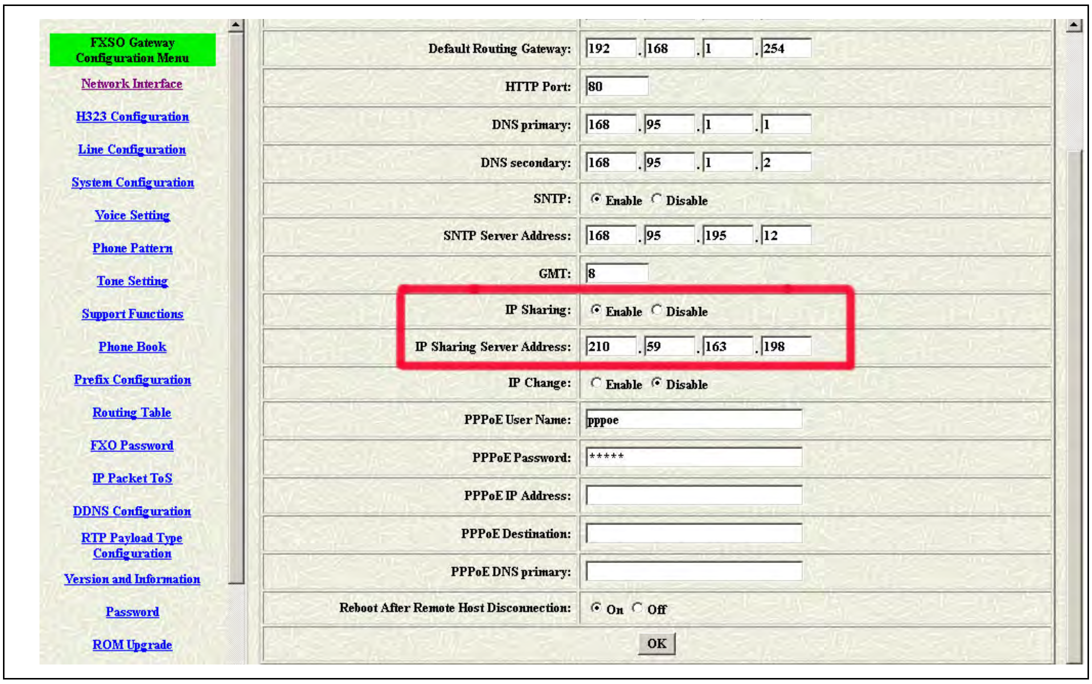


Figure 3.2: Enable the ip sharing function

1. Fixed IP – Please put the ip address of the ip sharing device in the IP Sharing Server Address table.
2. Dynamic IP – Just leave it if the IP sharing device is using PPPoE or DHCP protocol to connect with the internet.

3 The IP change function is only for the special usage

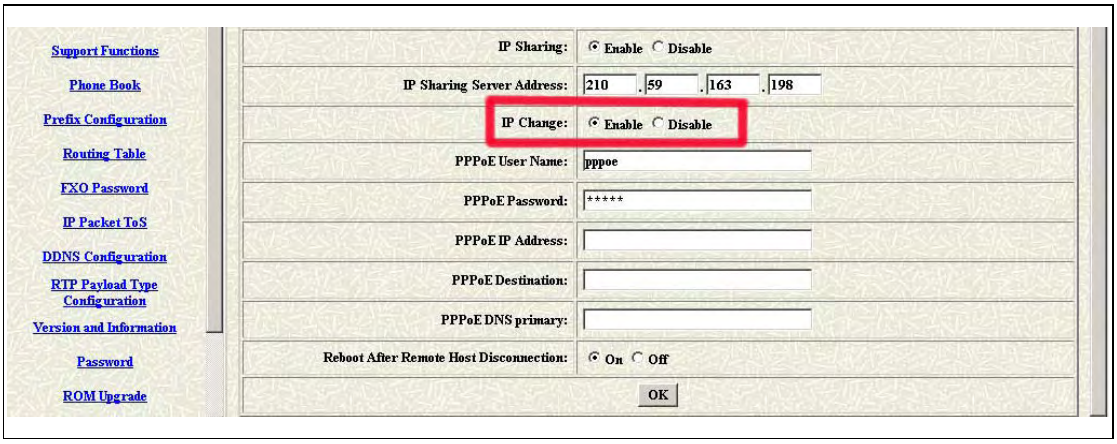


Figure 3.3: IP Change

- IP Change Function: The ip address of the IP Sharing Server Address will use what you configured in that table if you disable the IP Change function. It supports the static ip address. If user is using the dynamic ip address, this function has to be enabled.

If the wan port of the ip sharing device is using the dynamic ip address, the gateway couldn't get the ip address if the GK isn't from ours, too.

4 Please Commit it and Reboot the system if the configuration is finished.

This chapter is focus on the DMZ function of the IP Sharing Device, Please get another application from our web site.

3.2 Network Interface

User could configure the networking environment in this web page or change some default setting for networking. Please get more info from the following description. (see figure 3.4)

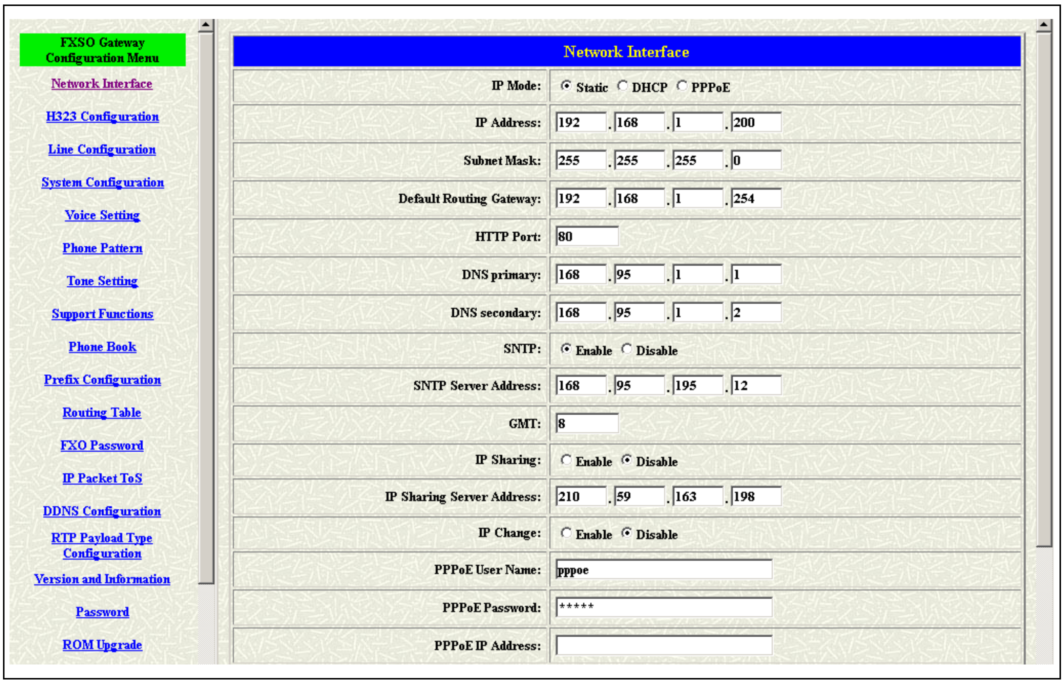


Figure 3.4: Network Interface

- IP Mode – Users could define the networking type for this gateway. It could support the Static, DHCP and PPPoE function.
- IP Address – Define the ip address for your networking if it is the fixed ip. Please get this info from your ISP.
- Subnet Mask – Define the mask address for your networking. Please get this info from your ISP.
- Default Gateway – Define the default gateway for your networking. Please get this info from your ISP.
- HTTP Port – This port is for the WEB configuration. The default port for the WEB is users could change it by this table.
- DNS primary – Users could define the primary DNS server address.
- DNS secondary – Users could define the primary DNS server address.

- SNTP – Enable the SNTP server registering function if user wants to get the correct time from the Command Line Interface.
- SNTP Server Address – Enter the correct ip address of the SNTP server or get the incorrect time from the Command Line Interface.
- GMT – Configuring the time area for the time display in the Command Line Interface.
- IP Sharing – Enable this function if the gateway is behind the IP sharing device.
- IP Sharing Server Address – Enter the WAN IP address of the IP sharing device if it is the fixed ip.
- IP Change – Enable this function if the WAN IP address of the IP sharing device is dynamic address.

The IP change function could support the GK from Asotel only. Please pay more attentions about this function if your IP sharing device is using the dynamic IP address.

- PPPoE User Name – Put the PPPoE connection account in this table. Please get this info from your ISP.
- PPPoE Password – Put the PPPoE connection password in this table. Please get this info from your ISP.
- PPPoE IP Address – After the connection success, this table will show you the IP address which the gateway got from the ISP.
- PPPoE Destination – After the connection success, this table will show you the default gateway address, which the gateway got from the ISP.
- PPPoE DNS primary – After the connection success, this table will show you the DNS ip address from the ISP.
- Reboot After Remote Host Disconnection – Enable this function will make the gateway restart automatically if the PPPoE connection is disconnected or the IP address was taken back by the ISP.

3.3 H323 Configuration

This page could support user configure the info about GK or change the mode from GK mode to Peer to Peer mode. Please get more detail info from below. (see figure 3.5)

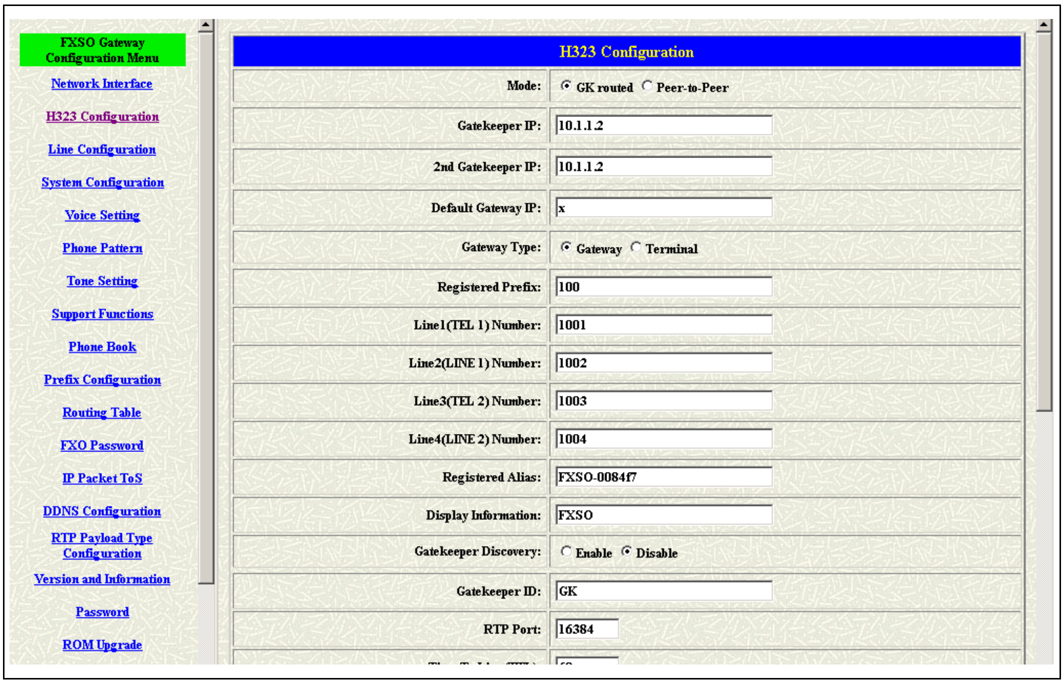


Figure 3.5: Network Interface

- Mode – Change the mode for this unit using.
  - GK routed : Users have to registered on the GK if users picked up this option.
  - Peer-to-Peer : It only supports the peer-to-peer mode and users have to define the phone book for this mode.
- Gatekeeper IP Address – Enter the GK ip if users pick up the GK routed mode. Support the GK mode only.
- 2nd Gatekeeper IP – It's for the back up using. Gateway will send RRQ to this GK if didn't receive the RCF from the First GK about one minute. Support the GK mode only.
- Default Gateway IP – All the calls will be routed to this destination if the destination couldn't be found in the Phone Book configuration. Support the

Peer-to-Peer mode only.

- ◆ Gateway Type – Pick up the type for gateway. Support the GK mode only.
  - Gateway : Could support one-stage-dialing function.
  - Terminal : Couldn't support one-stage-dialing function.
- ◆ Registered Prefix – The phone number for the GK registering. Support the GK mode only.
- ◆ Line1 (TEL 1) Number – Configure the number for the 1st FXS port.
- ◆ Line2 (TEL 2) Number – Configure the number for the 1st FXO port.
- ◆ Line3 (TEL 3) Number – Configure the number for the 2nd FXS port.
- ◆ Line4 (TEL 4) Number – Configure the number for the 2nd FXO port.
- ◆ Registered Alias – The name of this gateway for the GK registering. Support the GK mode only.
- ◆ Display Information – For the display name in the packets.
- ◆ Gatekeeper Discovery – Gateway will send the GRQ message and it will register on the GK if it had received the GCF message.

When users enable this function, the GK name is necessary for this. User could enable this function first and define the name of the GK. Gateway will send the GRQ message by broadcast if users define the IP address of GK is 255.255.255.255. If the gateway receive GCF message that's meaning the GK accept the request from gateway, so the gateway could register on that GK successfully who reply the GCF message.

- ◆ Gatekeeper ID – The name of the GK. It has used with the Gatekeeper Discovery function. Support the GK mode only.
- ◆ RTP Port – The UDP port for the voice sending. RTP ports support a range of the UDP. The line 1 is using UDP(RTP) 16384 and (RTCP) 16385. The line 2 is using UDP(RTP) 16386 and (RTCP) 16386….etc. This configuration is defining the start port for the RTP packets. Support the GK and Peer-to-Peer mode both. From 1024 to 65535.
- ◆ Time To Live(TTL) – The time for the registered confirm. Support the GK mode only. From 0 to 3600.
- ◆ Gatekeeper finding port – The port for the Gatekeeper Discovering function for this gateway. Support the GK mode only. From 1024 to 65535.
- ◆ Gatekeeper RAS Port – The GK registering port of this gateway. Support

the GK mode only. From 1024 to 65535.

- H225 RAS Port – The RRQ sending port of this gateway. Support the GK mode only. From 1024 to 65535.
- H225 Call Signal Port – The Call Signal Port sending of this gateway. Support the GK and Peer-to-Peer mode both. From 1024 to 65535.
- Destination H225 Call Signal Port – The destination Call Signal Port. Support the GK and Peer-to-Peer mode both. From 1024 to 65535.
- Allocate Port Range Start – The port range for this gateway. From 1024 to 19999.
- Allocate Port Range End – The port range for this gateway. From 1024 to 19999.
- Response Timeout – The call will be timed out if the call proceeding message didn't received from the remote side. Support the GK mode only. From 1 to 200.
- Connection Timeout – The call will be timed out if the connect message didn't received from the remote side. Support the GK mode only. From 1 to 20000.
- H.235 Security Token – Support H235 security password for the registration.

3.4 Line

The Line configuration will show the status of the registrations and the ports. It includes the hunt group, hotline, and no answer forward configuration. Press the Line configuration button to enter configuration table (see figure 3.6)

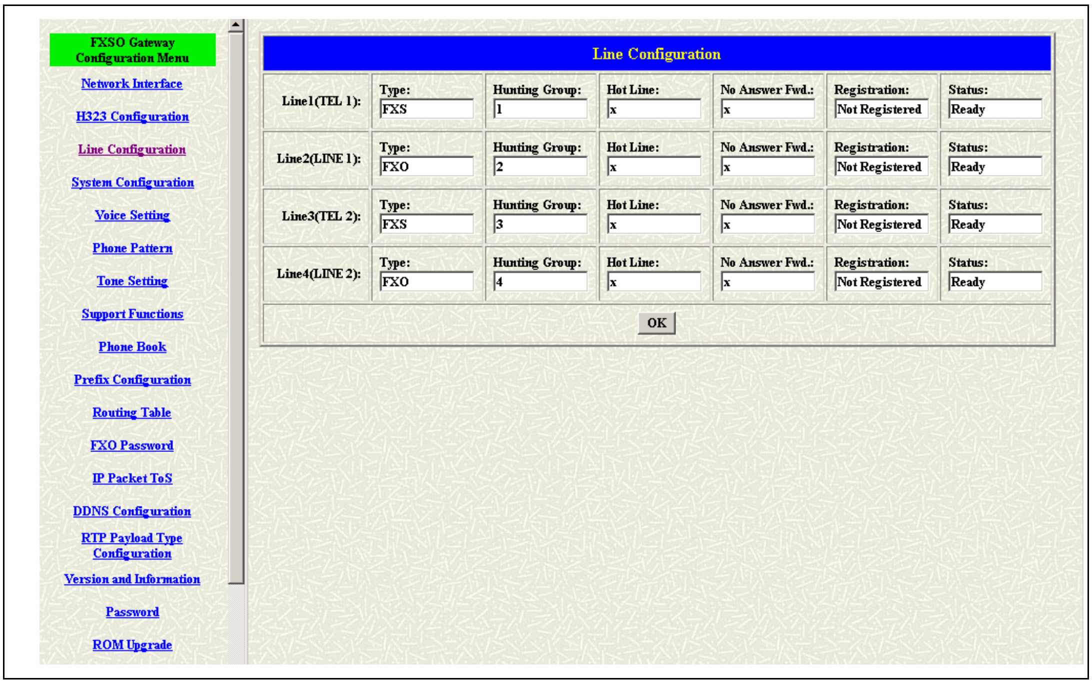


Figure 3.6: Line Configuration

- Type – Show the type of this port. There are only two types of this gateway. One is FXS type another is FXO type. It couldn't be changed.
- Hunting Group – Define the group number of this port. When the port is busy, the call could be transferred to another port in the same group. Only the same type could be configured in the same group.
- Hotline – Enable or Disable the hotline mode. The hotline mode will be enabled if you enter the hotline number. The default setting is disabled.
- No Answer Forward – When the port didn't answer the call, this call will be forwarded to the number you configured. This is only for the E164 number or the phone number you want to transfer.

- Registration – To show the gateway registered on the GK or not.
- Status – To show the port is busy or ready.

3.5 System Configuration

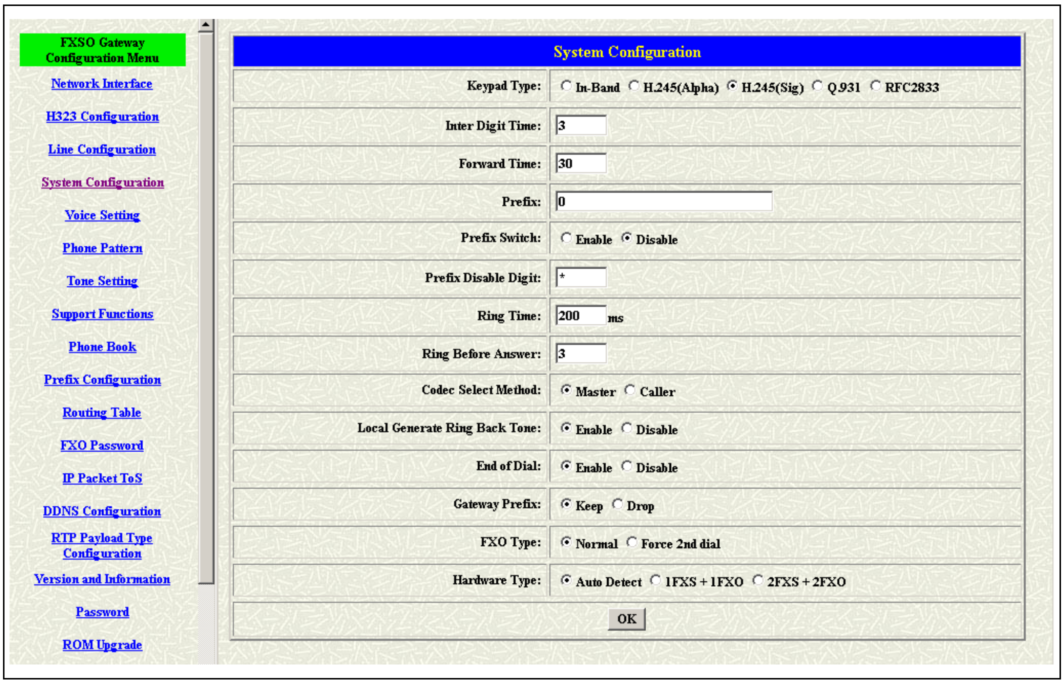


Figure 3.7: System Configuration

- Keypad Type – User could define the keypad for the DTMF sending.
  - In-Band : The DTMF signal sending by RTP.
  - Out-Band : The DTMF signal sending not by RTP. Including the H.245(Alpha), H.245(Signal), Q.931 and RFC 2833.
- Inter Digit Time – The call will be sent out if user didn't enter the digits after this timer. From 1 to 10.
- Forward time – It supports the No Answer Forward function. If users configure it for 10, the call will be forwarded when it rings about 10 seconds. From 5 to 65535.
- Prefix – This will be added in the first digits of the numbers that users had dialed.
- Prefix Switch – Users could enable or disable this function.
- Prefix Disable Digits – Users could enter this digits for disable this function in this dialing.

The Prefix, Prefix Switch and Prefix Disable Digits is a function. Users have to configure these three tables if this function is needed.

- Ring Time – It for the ring detection from the PSTN. The ring detection will be failed if users configure it too long.
- Ring Before Answer – This will help the users to answer the calls from PSTN into this gateway quickly. The call will be connected by one time ring if users configure this for 1. From 1 to 10.
- Codec Select Method – This could support that the codec will follow the MSD (MasterSlaveDetermination) or the caller side.
  - Master : Follow the result from the MasterSlaveDetermination.
  - Caller : Follow the caller side.

It's only for the special requirement. Please contact with your vendor before you configure this.

- Local Generate Ring Back Tone – To enable or disable the ring back tone generation from the local side.
- End of Dial – To enable or disable the end of dial function. This function key will be the digit "#".
- Gateway Prefix – To keep or drop the prefix number of this gateway. This only support the Gateway type in the GK routed mode. Please get more detail useful as below :

The gateway could keep or drop the prefix number if the call is coming from the ip side and bring the prefix number of this gateway. The usage of this command is about the one-stage-dialing function. Of course it must work with the Routing table.

Source	Gateway Prefix	Receive	Transfer
10012345	Keep	10012345	10012345
10012345	Drop	10012345	12345

If the incoming call (From IP side) with the number is 10012345, and the prefix number of the destination gateway is 100. The gateway will receive 10012345. But the number it transfer will be 10012345 (Gateway Prefix Keep). The transfer number will be 12345 if the Gateway Prefix function is enabled.

- FXO type – Users could configure all the calls need the 2^{nd} stage dialing or not.
  - Normal : The 1^{st} or 2^{nd} stage dialing will depend on the dialing plan from the users. If user dial the number of the FXO port, that will be the 2^{nd} stage dialing.
  - Force 2^{nd} dial : Every calls will need the 2^{nd} stage dialing type.
- Hardware Type – This will show the hardware detection type.

3.6 Voice Setting

Users could configure the voice codec or gain level in this web page. Please get more detail info from the following description. (see figure 3.8)

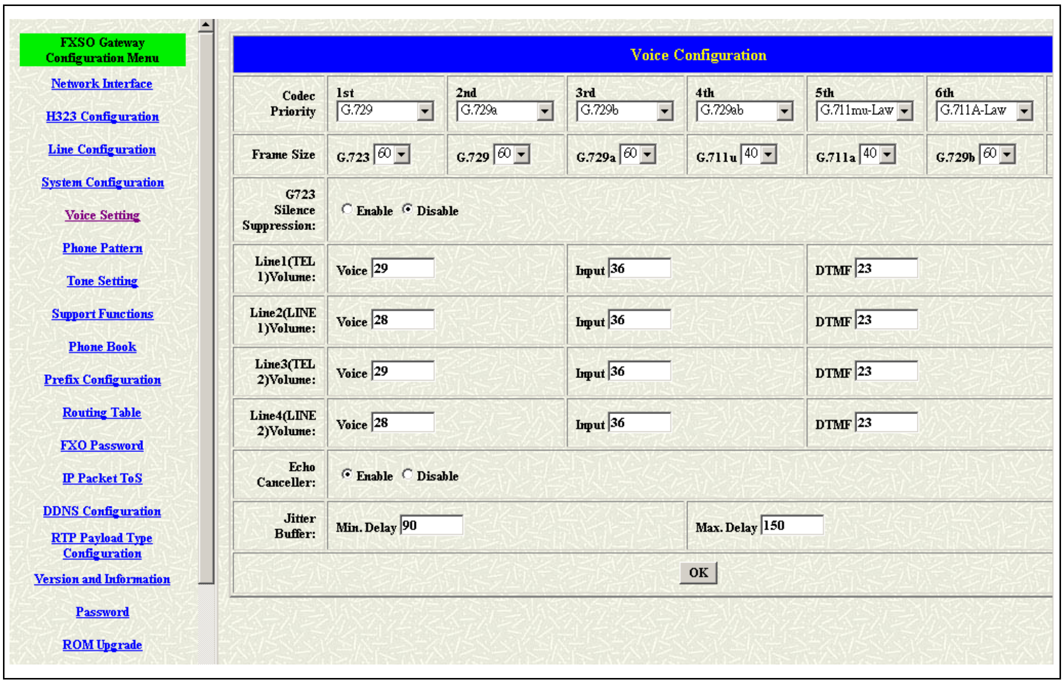


Figure 3.8: Voice Setting

- Codec Priority – This could help users configure the codec priority for using. Please pay more attentions about the firmware you use. If the firmware only support G.729 series codec, the G.723 codec in this table will be useful.
- Frame Size – To configure the packet size for the codec that users want.
- G723 Silence Suppression – To enable the VAD and CNG function for the G.723 codec.
- Line1 (TEL 1) Volume – To configure the output gain (voice), input gain (input) and DTMF gain (DTMF) of the first FXS port.
- Line2 (LINE 1) Volume – To configure the output gain (voice), input gain (input) and DTMF gain (DTMF) of the first FXO port.
- Line3 (TEL 2) Volume – To configure the output gain (voice), input gain (input) and DTMF gain (DTMF) of the second FXS port.

- Line4 (LINE 2) Volume – To configure the output gain (voice), input gain (input) and DTMF gain (DTMF) of the second FXO port.
- Echo Canceller – To enable or disable the echo cancellation function.
- Jitter Buffer – TO configure the Min or Max delay for the Jitter Buffer. The min is from 0ms and the max is 150ms.

3.7 Phone Pattern

This WEB will show the tone generation value from this unit. Please don't configure it if you got some correct value for sure. (see figure 3.9)

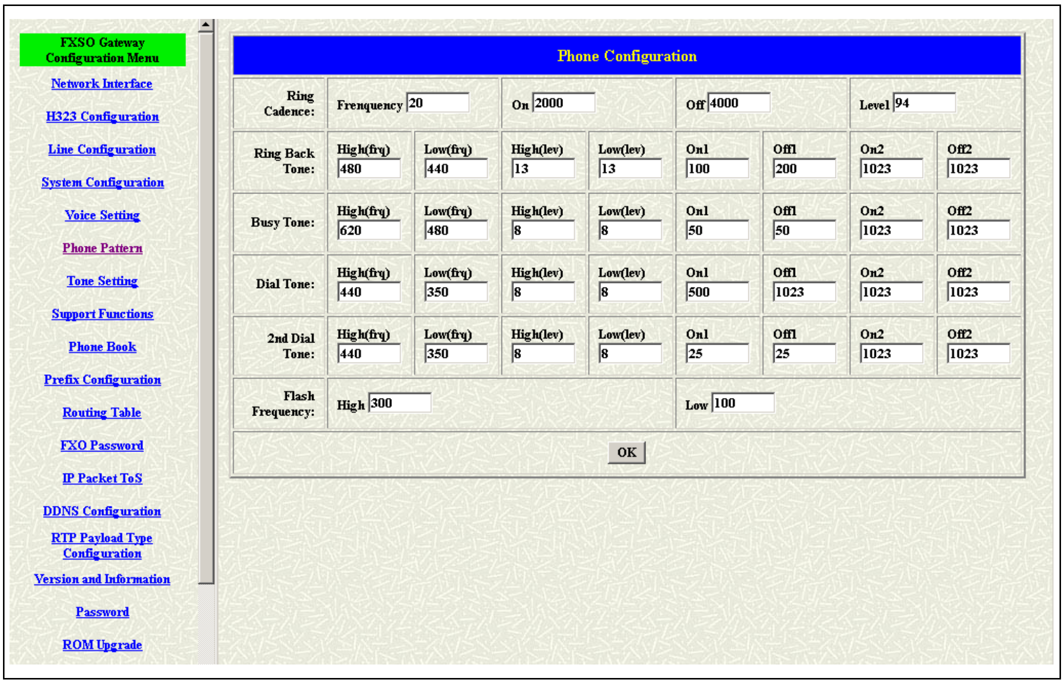
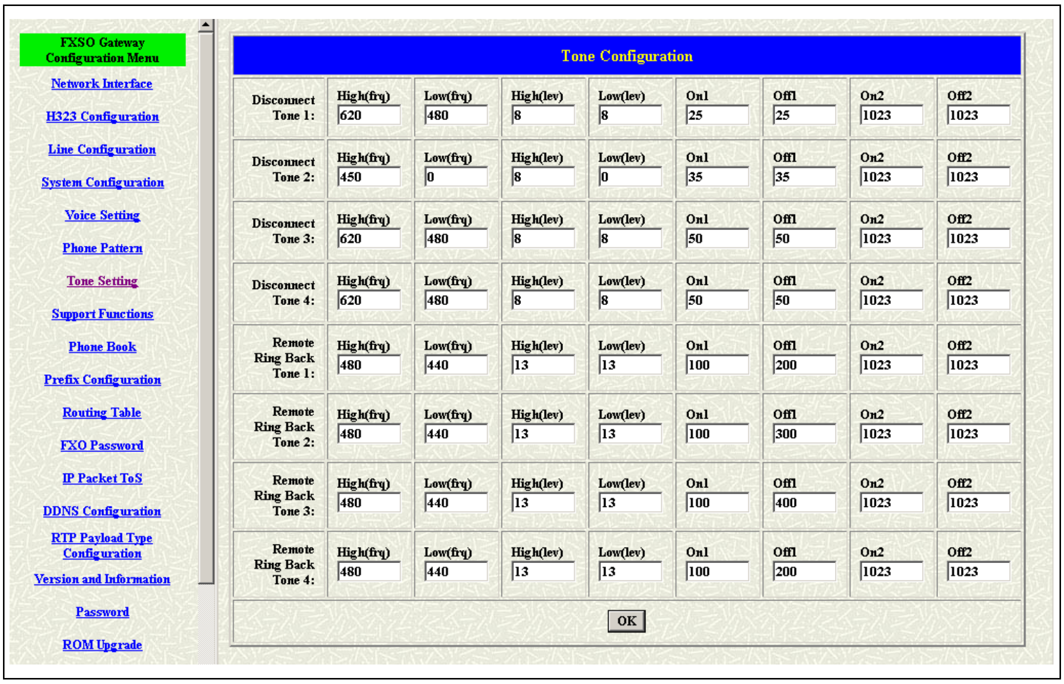


Figure 3.9: Phone Setting

- Ring Cadence – TO configure the frequency and cadence for the local ring tone. The cadence is for ms.
- Ring Back Tone – To configure the value of the local ring back tone generation
- Busy Tone – To configure the value of the local busy tone.
- Dial Tone – To configure the value of the local dial tone.
- 2nd Dial Tone – To configure the value of the local 2nd dial tone.
- Flash Frequency – To configure the High and Low frequency for the Flash sending.

3.8 Tone Setting

Users could configure the tone pattern of the gateway in this page. If the disconnect tone from PSTN side is match the one of the Busy tone table, the call will be dropped. (see figure 3.9)



Tone Configuration

	High(frq)	Low(frq)	High(lev)	Low(lev)	On1	Off1	On2	Off2
Disconnect Tone 1:	620	480	8	8	25	25	1023	1023
Disconnect Tone 2:	450	0	8	0	35	35	1023	1023
Disconnect Tone 3:	620	480	8	8	50	50	1023	1023
Disconnect Tone 4:	620	480	8	8	50	50	1023	1023
Remote Ring Back Tone 1:	480	440	13	13	100	200	1023	1023
Remote Ring Back Tone 2:	480	440	13	13	100	300	1023	1023
Remote Ring Back Tone 3:	480	440	13	13	100	400	1023	1023
Remote Ring Back Tone 4:	480	440	13	13	100	200	1023	1023

Figure 3.10: Phone Setting

- Disconnect Tone – To configure the frequency, level gain and on/off time for the busy tone from PSTN side. The busy tone supports 4 tables.
- Remote Ring Back Tone –To configure the frequency, level gain and on/off time for the ring tone from PSTN side. The gateway won't connect the calls if the ring tone value is incorrect.
- Dial Tone – To configure the frequency, level gain and on/off time for the dial tone.
- Ring Back Tone – To configure the frequency, level gain and on/off time for the dial tone.

3.9 Support Functions

This gateway support the FAX over IP, fast start function and others function. Please enable or disable these in the Support WEB page. (see figure 3.10)

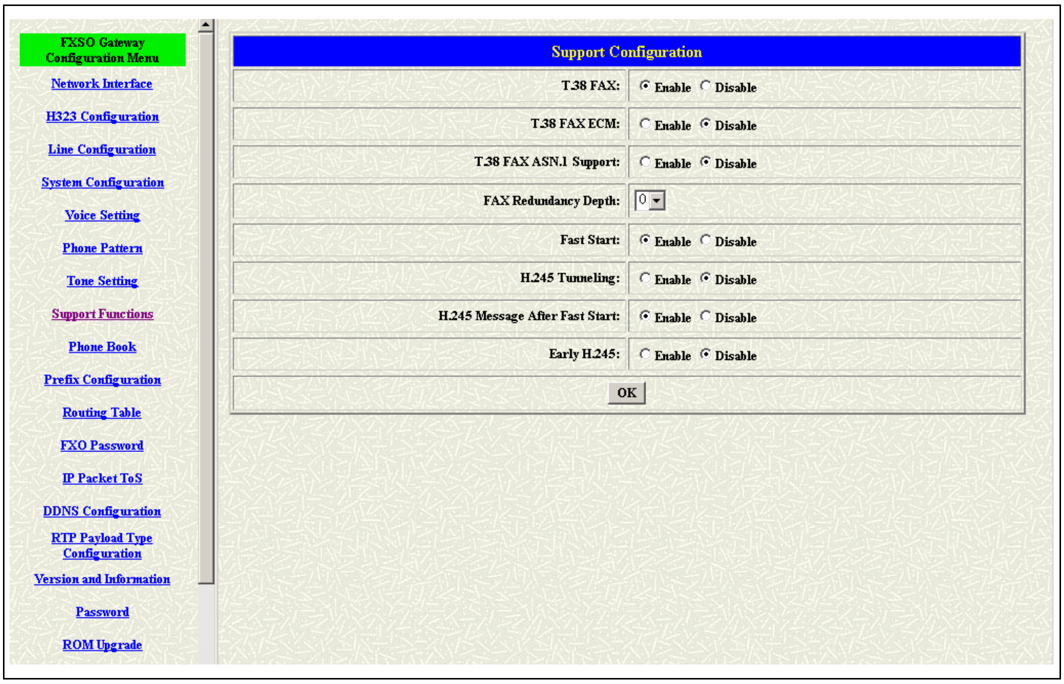


Figure 3.11: Support Functions

- T.38 FAX – Enable this function to support the FAX function.
- T.38 ECM – This function could support the error correction mode during the high-speed function.
- T.38 FAX ASN.1 Support – Support the ASN.1 function.
- Fax Redundancy depth – This support function could make the data for the FAX sending for twice. But this will take more bandwidth.
- Fast Start – Enable this function will support the Fast start function.
- H.245 Tunneling – Enable or disable the Tunneling support.
- H.245 after faststart – Enable or disable the H.245 after fast start function.
- Early H.245 –

3.10 Phone Book

The Phone Book configuration is only support the gateway in Peer-to-Peer mode. Please refer the chapter 2 about the Peer-to-Peer mode. (see figure 3.11)

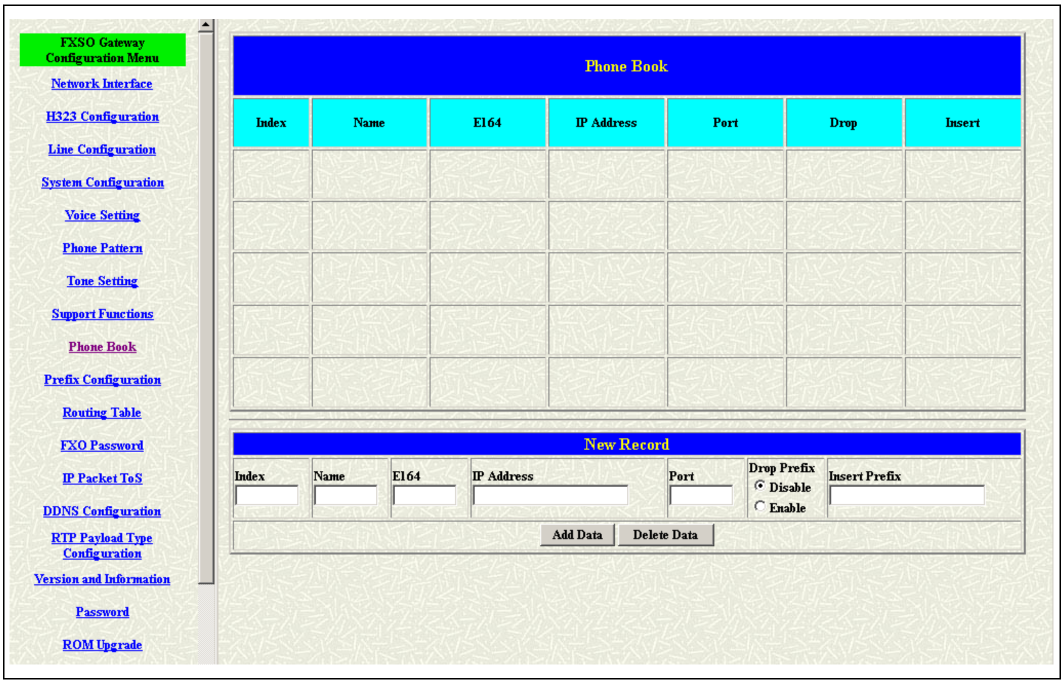


Figure 3.12: Phone Book

- Index – The list number of the Phone Book.
- Name – The name for this contact number.
- E164 – The dialing number for the calling side.
- IP Address – The destination IP address for this phone number.
- Port – The call signal port of the destination.
- Drop Prefix – Support the drop function. Enable is for enable this drop function; Disable is for disable this drop function. The Drop Prefix will drop the E164 number, which you had configured in the E164 table.
- Insert Prefix – Support the insert digits function.

1. It will be the drop function if user enable the Drop Prefix function and put nothing into the Insert Prefix table.
2. It will be the insert function if user disable the Drop Prefix function and put the digits into the Insert Prefix table.
3. It will be the replace function if user enable the Drop Prefix function and put the digits into the Insert Prefix table.

- Add Data – Press this button if users fill the entire information table above.
- Delete Date – If users want to delete the record from the table, enter the index number first and press this button. The record will be deleted.

3.11 Prefix Configuration

The Prefix function is using the drop and insert function (see figure 3.8).

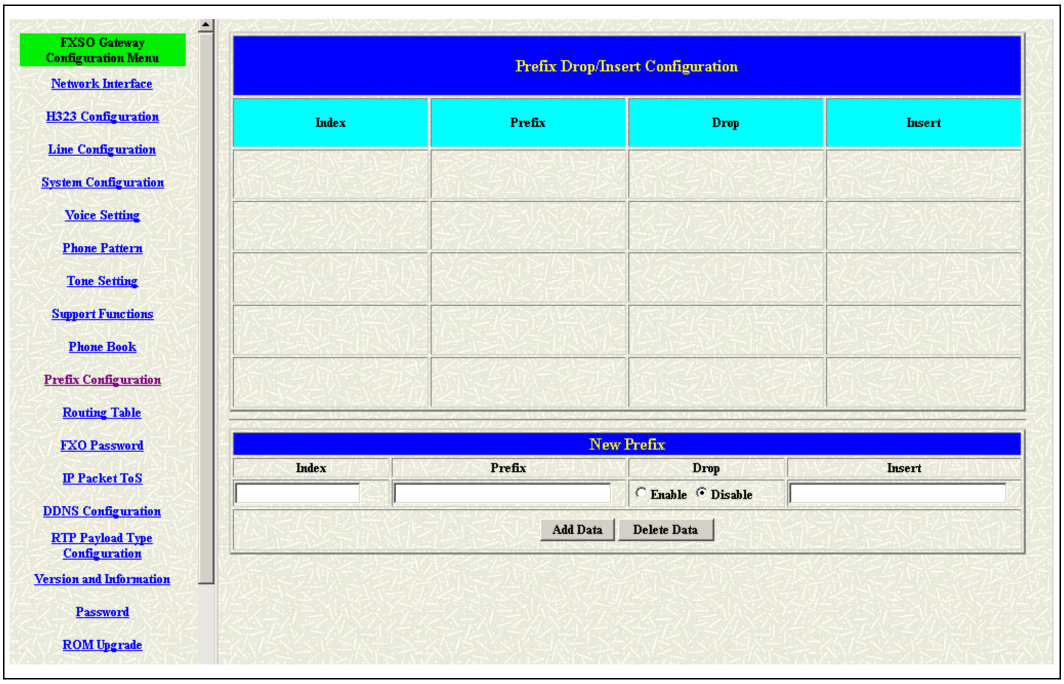


Figure 3.13: Prefix Configuration

- Index – The list number of the Phone Book.
- Prefix – The prefix number of the whole numbers that could be into this gateway
- Drop – The drop function. Enable this function by the Enable button; Disable this function by the Disable button.
- Insert – The insert function. Users could enter the digits that you want to insert in this number.
- Add Data – Press this button if users fill the entire information table above.
- Delete Date – If users want to delete the record from the table, enter the index number first and press this button. The record will be deleted.

This function is just like the Phone Book configuration. But it will make the drop and insert function in the GK routed mode. All the numbers into this gateway will check out the prefix table first and find out the destination in the Routing Table.

There is an example about the configuration, please follow up these steps.

1 Press the Prefix Configuration button to enter the configuration table (see figure 3.13)

2 Enter the index number. Put the prefix numbers you will dial in the prefix table, enable (disable) the drop function and enter the numbers you want to insert (see figure 3.14)

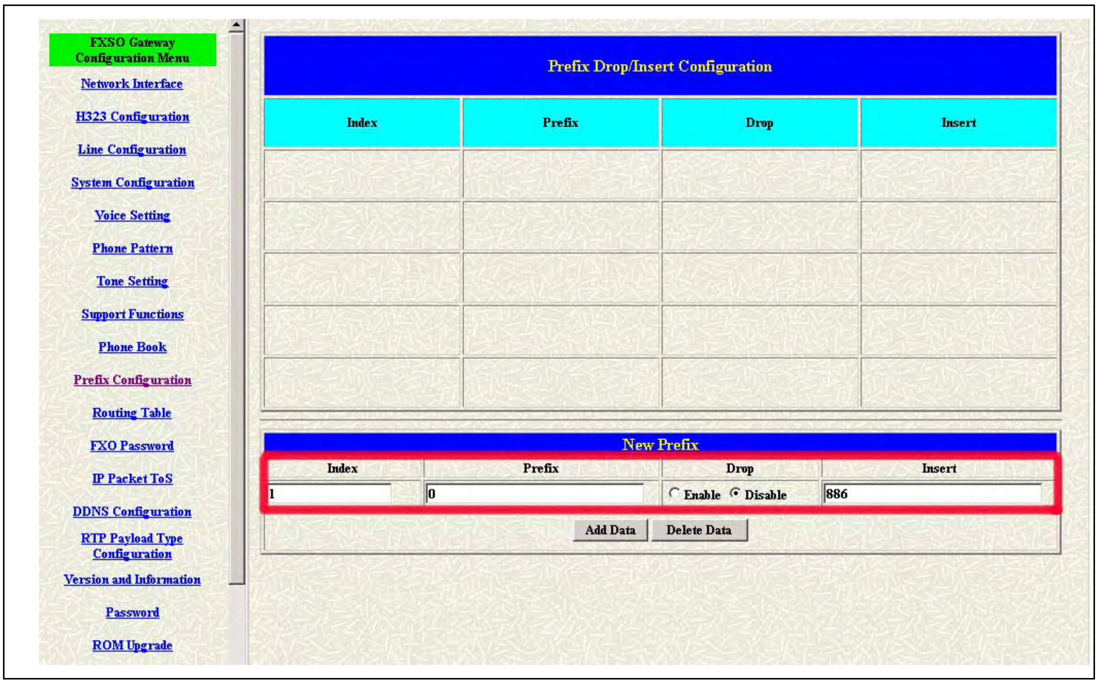


Figure 3.14: Configure the Prefix Table

The usage is as same as the drop, insert function of the Phone Book.

Input (Prefix)	Drop	Insert	Output
100	Disable	X	100
200	Disable	0	0200
300	Enable	X	X

400	**Enable**	**500**	**500**

3 Press the Prefix Configuration button to reload the configuration table (see figure 3.15)

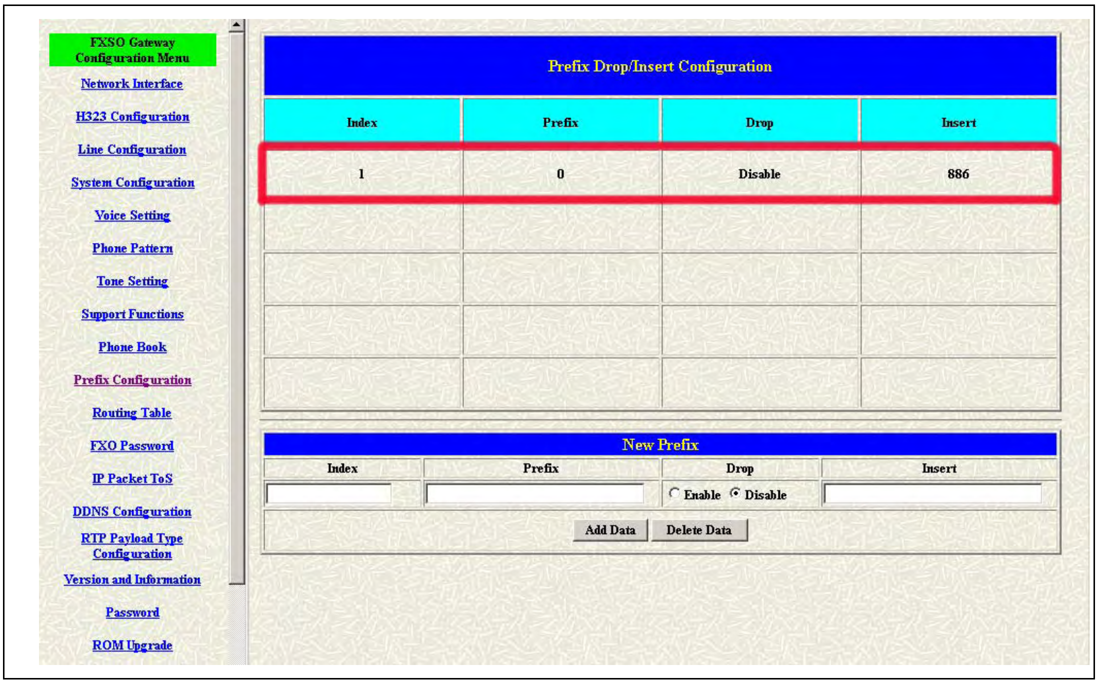


Figure 3.15: Show the added table

4 Please Commit it and Reboot the system if the configuration is finished.

3.12 Routing Table

Routing Table is a rule to define the destination of the calls you make. You could define the rules by the number you dial or by the ports. The Routing Table button will show you the configuration table. (see figure 3.16)
In fact, there are three directions of the incoming calls (from IP, FXS and FXO side). The explanation of the default routing is as below:

The location with the incoming calls	The location with the destination	The explanation
IP (Default)	Fxs	The destination will be the FXS port when the calls from the IP side without any define rules.
Fxs (Default)	IP	The destination will be the IP side when the calls from the FXS port without any define rules.
Fxo (Default)	IP	The destination will be the IP side when the calls from the FXO port without any define rules.

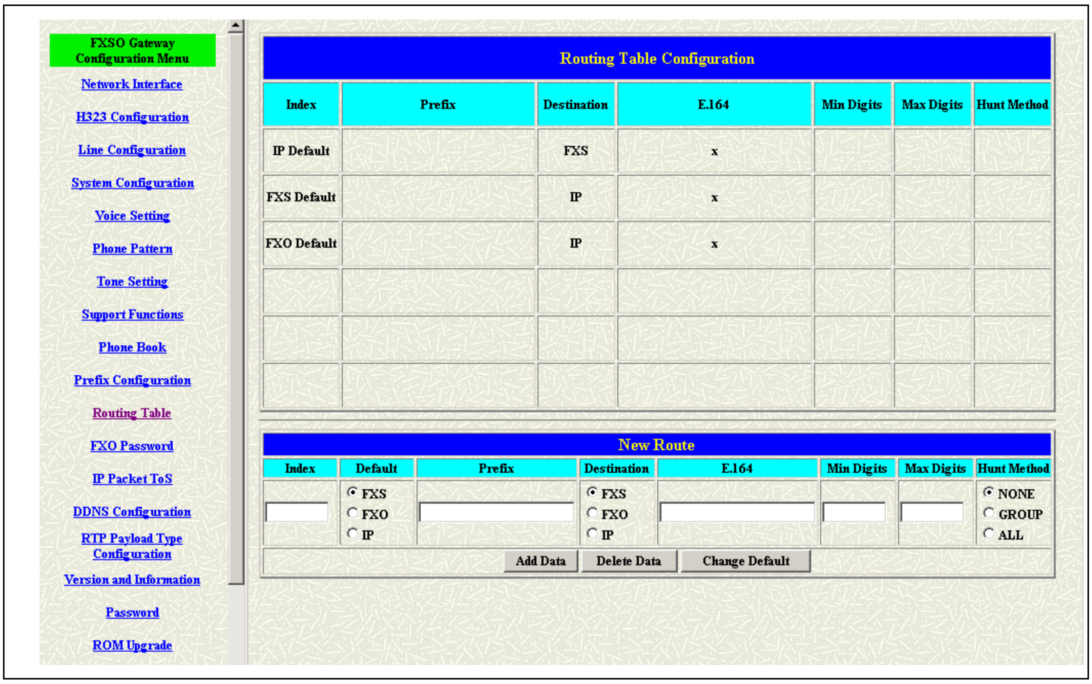


Figure 3.16: Routing Table Configuration

- Index – The list number of the Route Table.
- Default – For change the default setting. Users have to pick the direction for the default setting changed.
- Prefix – The prefix number for the dialed digits. The call will be followed this route table if the prefix number was matched.
- Destination – To decide the destination for this route table.
- E.164 – The E.164 number of the destination.
- Mini Digits – The mini digits requirement for this route table.
- Max Digits – The max digits requirement for this route table.
- Hunt Method – Enable the Hunt Group function and pick up the hunt type.
  - NONE : Disable the Hunt Group function.
  - GROUP : The Hunt Group function will working for the same group. User could configure the group in the Line Configuration table.
  - ALL : The Hunt Group will working for the same type.

None – Disable this function
Group – The call will search other ports to be the destination with the same group if the origin destination is busy.
All – The call will search other ports to be the destination with the same type if the origin destination is busy.
About the Group setting, Please get more info from the Line Configuration.

- Add Data – Add a new record for the route table.
- Delete Data – Delete a record for the route table.
- Change Default – Change default route table.

3.12.1 Change the default routing

Please follow up the steps if you want to change the default routing:

1 Pick up the side for the incoming calls and define the destination of this side. Press the Change Default to save the data.

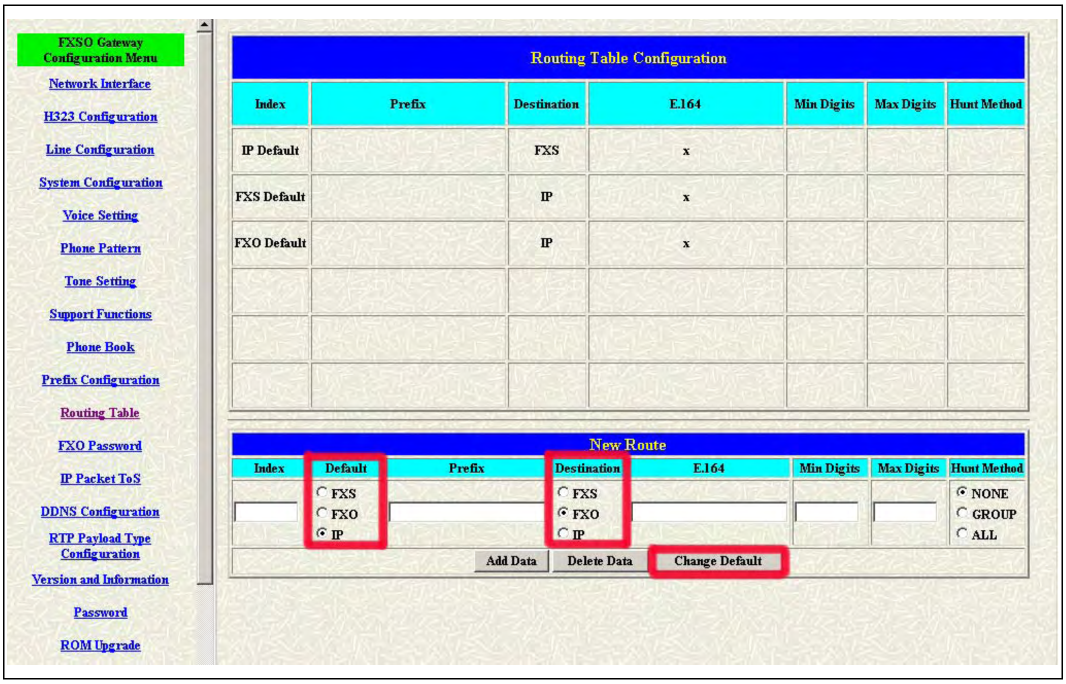


Figure 3.17: Change The Default Setting

2 The default setting is changed after you press the Change Default button. Please press the Routing Table button again to show the new setting.

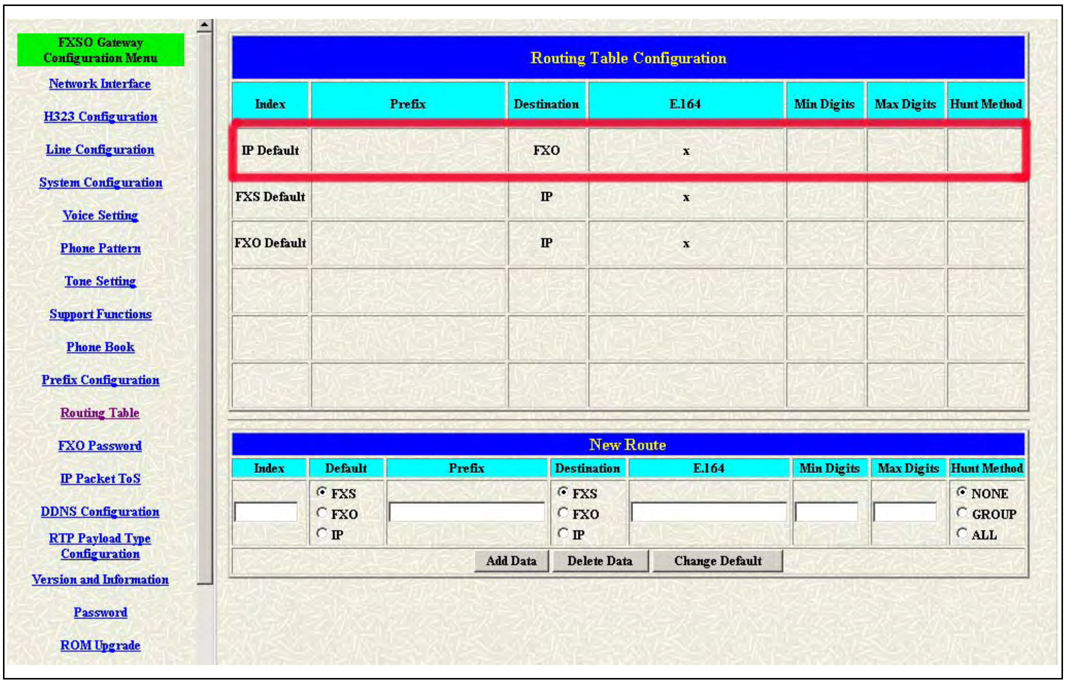


Figure 3.18: The Default Setting Changed

3 Please Commit it and Reboot the system if the configuration is finished.

3.12.2 Add a new Routing Table

1 The default setting is changed after you press the Change Default button. Please press the Routing Table button again to show the new setting.

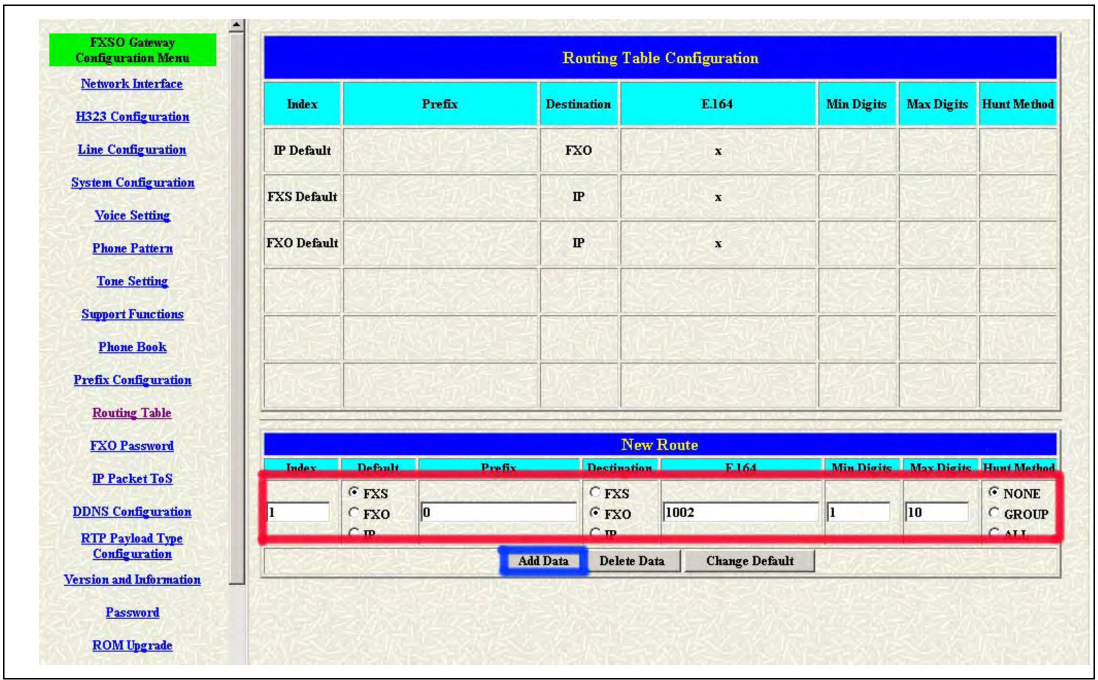


Figure 3.18: The Default Setting Changed

2 Press Add Data button to save the configuration and press the Routing Table button again to reload the configuration.

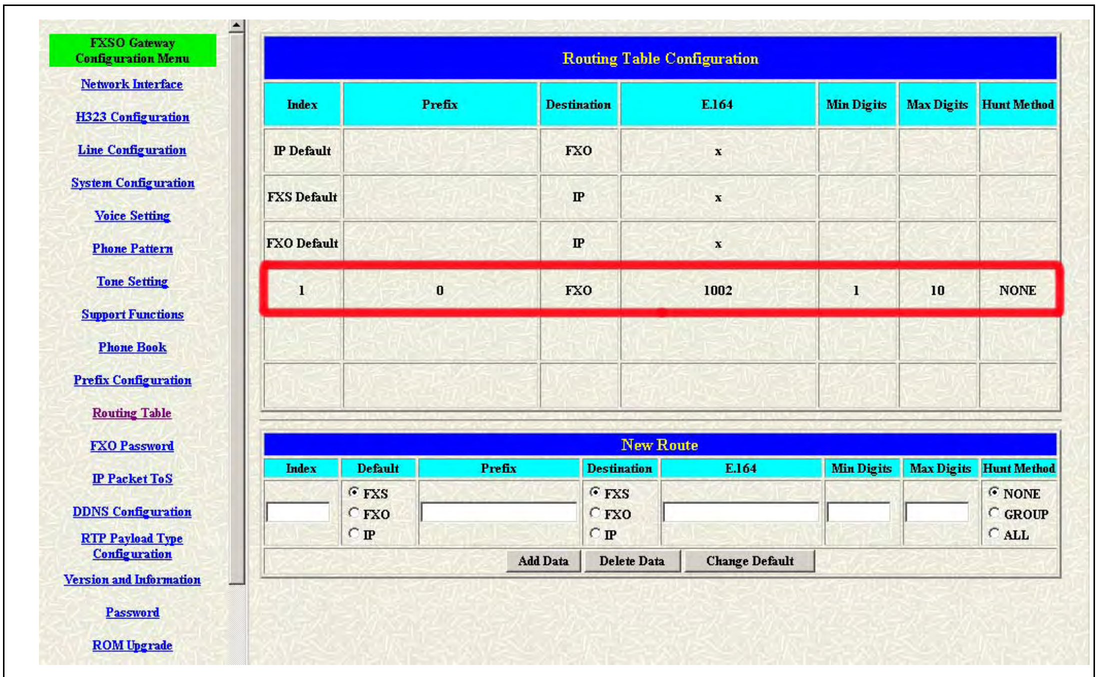


Figure 3.19 New Special Routing

The explanation of figure 3.19 is as below:

When the user dial 0 with the first digit of the numbers (from FXS side), or from FXO and IP side. And the numbers you dial is between 1 and 10 digits. If this call matches the rule, it will be transferred to the FXO port whose E164 number is 1002.

3 Please Commit it and Reboot the system if the configuration is finished.

3.13 FXO Password

You will get the IVR if you make calls from PSTN side or from IP side. The IVR will ask you the password you set, and you could make other calls to IP side or PSTN side if the password you type is correct. Please press the FXO Password button to configure the password (see figure 3.20)

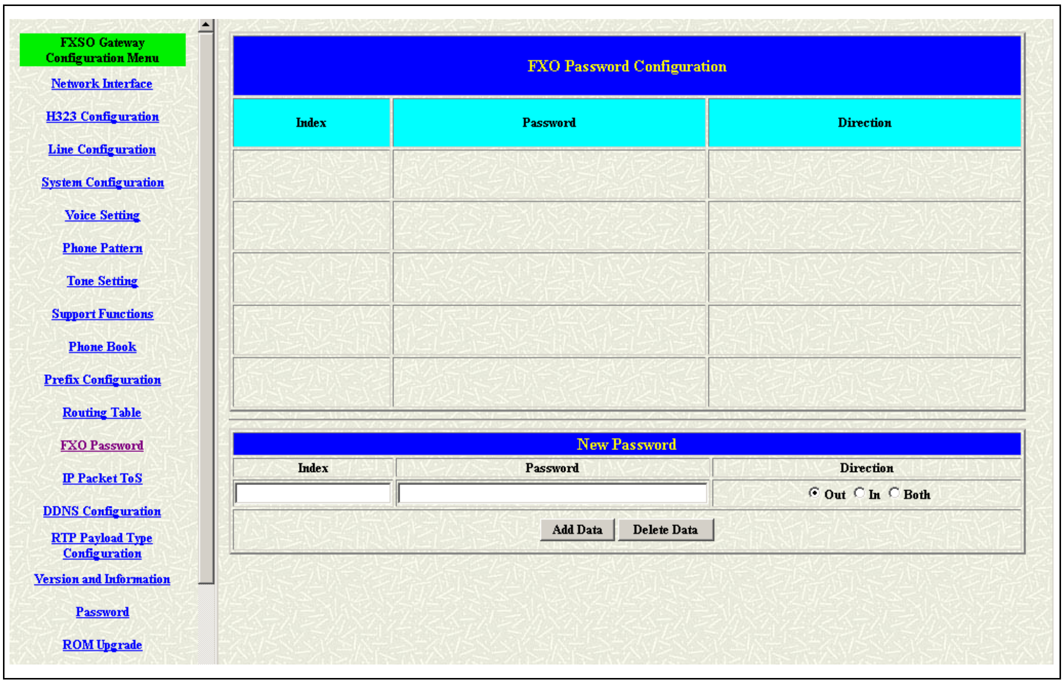


Figure 3.20: FXO Password

- Index – The number of this table.
- Password – The password you set.
- Direction – Playing the IVR to PSTN, IP or both side.

(Out – The calls from IP side ; In – The calls from PSTN side)

3.14 IP Packet ToS

The Type of Service should be worked with the network router. The router will check all the packets if it support the TOS function. There is a field in the packet for the TOS value. This WEB is for users to configure these values to make the packets with the correct values for the TOS service from the gateway. (see figure 3.21)

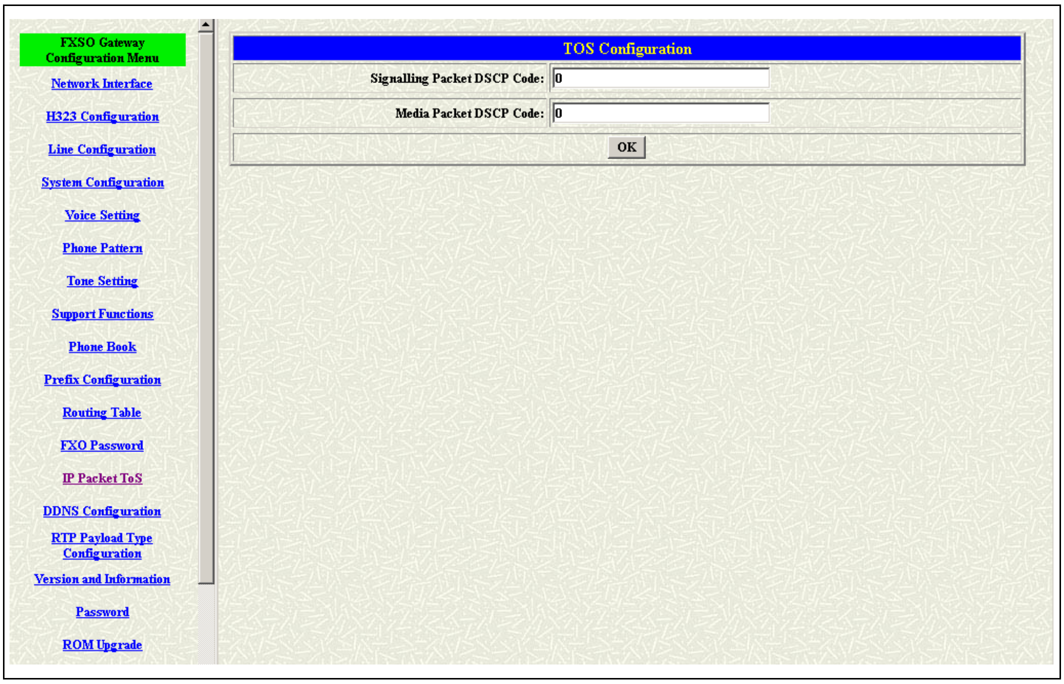


Figure 3.21: TOS Configuration

According to the RFC 1349 document, the TOS value as following :

1000 – minimize delay
0100 – maximize throughput
0010 – maximize reliability
0001 – minimize monetary cost
0000 – normal service

These values are the Binary format. Please change to the Decimal and put these values in to the correct table.

3.15 DDNS Configuration

This version of firmware supports DDNS function. Before using this function, please have a DDNS account and some info from your DDNS server. Press the DDNS Configuration button to configure it. (see figure 3.22)

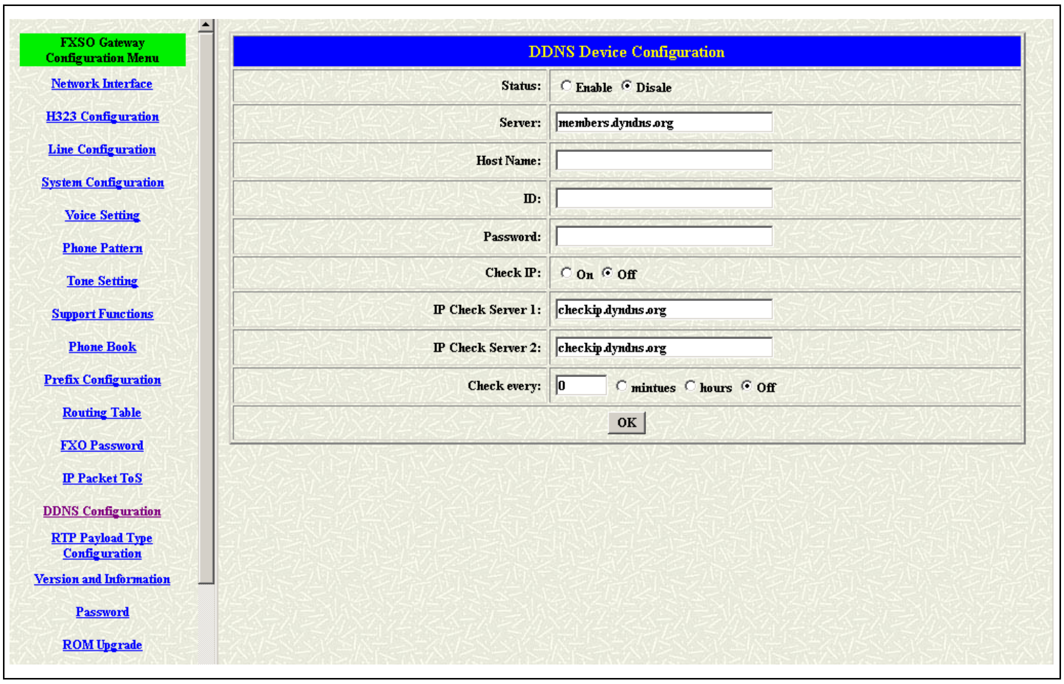


Figure 3.22: DDNS Configuration

- Status – To enable or disable this function.
- Server – Enter the server address of your DDNS server.
- Host Name – Your DDNS address.
- ID – Your account.
- Password – Your password.

Please get this info from your DDNS server.

- Check IP – Enable or Disable the IP Check function.
- IP Check Server – Will check the endpoints ip address.
- Check Every – The endpoint will check the server after a period.

3.16 RTP Payload Type Configuration

There are more types for the RTP Payload. This web page could support users define the Payload Type for some special payload. (see figure 3.23)

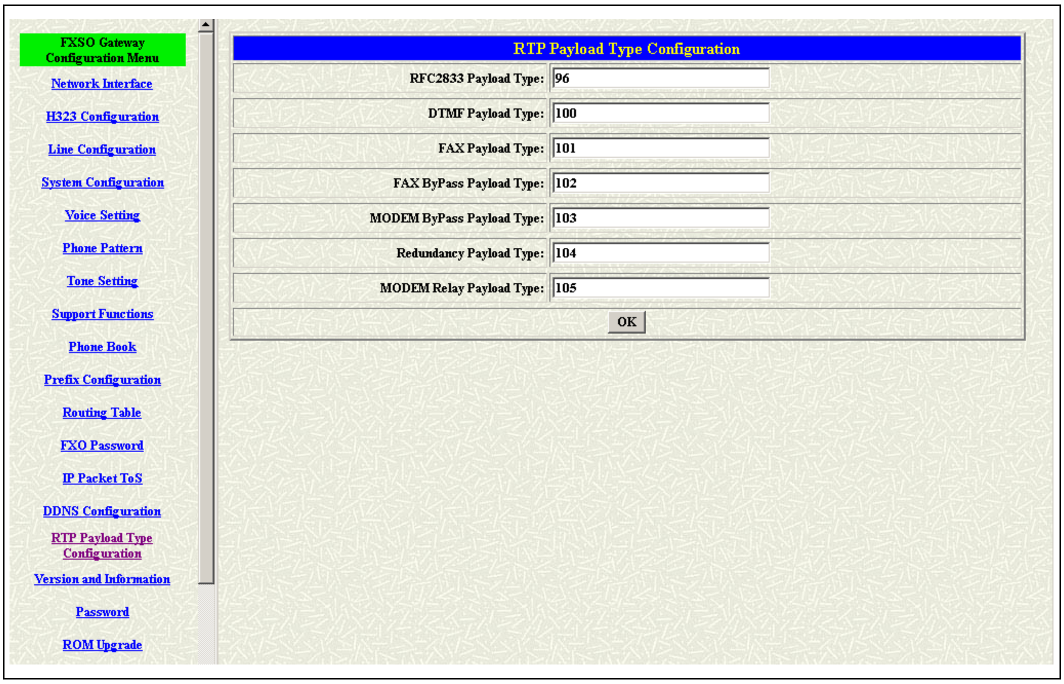


Figure 3.23: RTP Payload Type Configuration

- RFC2833 Payload Type – To define the payload type for RFC2833 type.
- DTMF Payload Type – To define the payload for the DTMF type.
- FAX Payload Type – To define the payload for the FAX type.
- FAXByPass Payload type – To define the payload for the FAX by Pass type.
- MODEMByPass Payload Type –To define the payload for the Modem by Pass type. (This is no use for the hardware as so far.)
- Redundancy Payload Type – To define the payload for the Redundancy type.
- MODEMRelay Payload Type – To define the payload for the FAX by Pass type. (This is no use for the hardware as so far.)

Please contact with your vendors if you want to configure these values.

3.17 Version and Information

Users could get more detail about the software version for all the parts in this web page. (see figure 3.24)

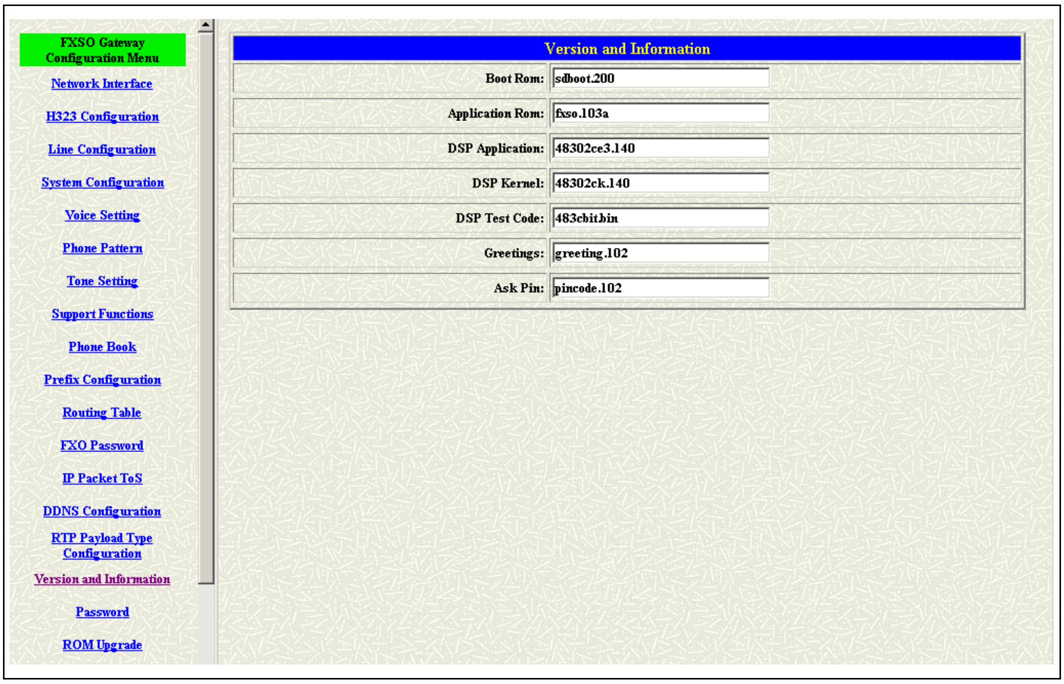


Figure 3.24 Version and Information

- Boot Rom – The version of the Boot Rom layer.
- Application Rom – The version of the Application Rom layer.
- DSP Application – The version of the DSP Application Rom layer.
- DSP Kernel – The version of the DSP Kernel layer.
- DSP Test Code – The version of the DSP Test Code layer.
- Greeting – The version of the Greeting file.
- ASK Pin – The version of the ASK Pin file.

3.18 Password

There are two-login accounts in this unit. One is the account root another is administrator. The default setting for these two accounts are empty. Users could define the passwords for these two accounts. Please get more info from the following description. (see figure 3.25)

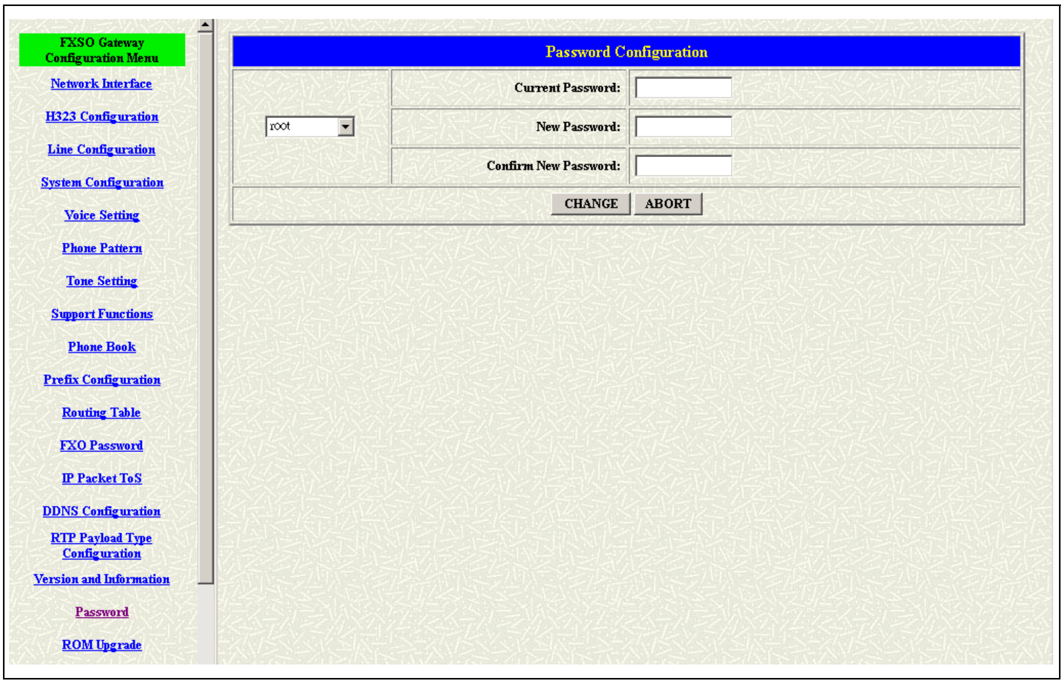


Figure 3.25 Password

- root – The password for the root account.
- administrator – The password for the administrator account. This account couldn't upgrade the 2M and boot rom file.
- Current Password – Enter the original password for the account.
- New Password – Enter the new password for the account.
- Confirm New Password – Enter the new password again.
- Change – This button will make the configurations saved and next time login will need the new password.
- Abort – Abort the configuration of the password changing.

3.19 ROM Upgrade

User could update the firmware just by the web configuration interface. There are two types for the upgrading procedure. One is using the TFTP server, another is using the FTP server. Please follow the step to update the gateway firmware version. (see figure 3.26)

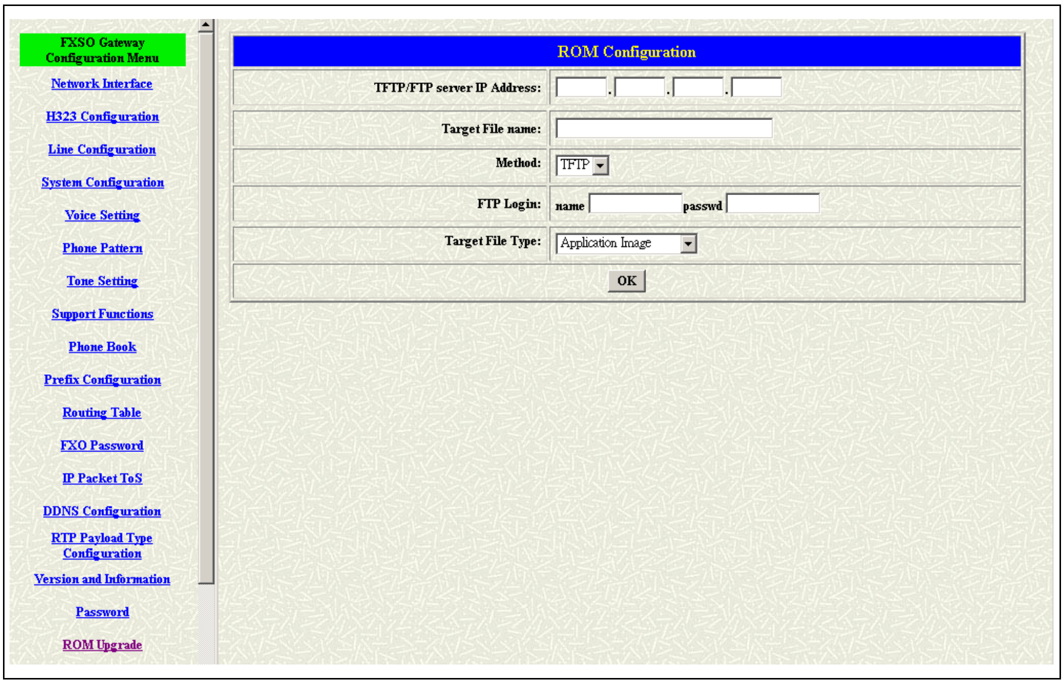


Figure 3.26 ROM Upgrade

- TFTP/FTP server IP Address – Put the ip address of the TFTP or FTP server in this table.
- Target File name – Put the target file name in this table.
- Method – There are two upgrade methods for the upgrade procedure. One is TFTP and another is FTP. Please change to correct method for the upgrading.
- FTP Login – Please enter the login name and password for the FTP upgrade method. This is necessary if user change the method to the FTP.
- Target File Type – Please pick up the correct file type for upgrading. If the file name and the file type is unconformable, the upgrade procedure will be failed.

◆ OK – Press the OK button if all the info above are correct. The unit will start to download the firmware file from the TFTP or FTP and write to the flash after the downloading.

Updating the firmware by the FTP server

1 Pick up the "Rom Upgrade" button to enter the upgrading web page and switch to the FTP method. (see figure 3.27)

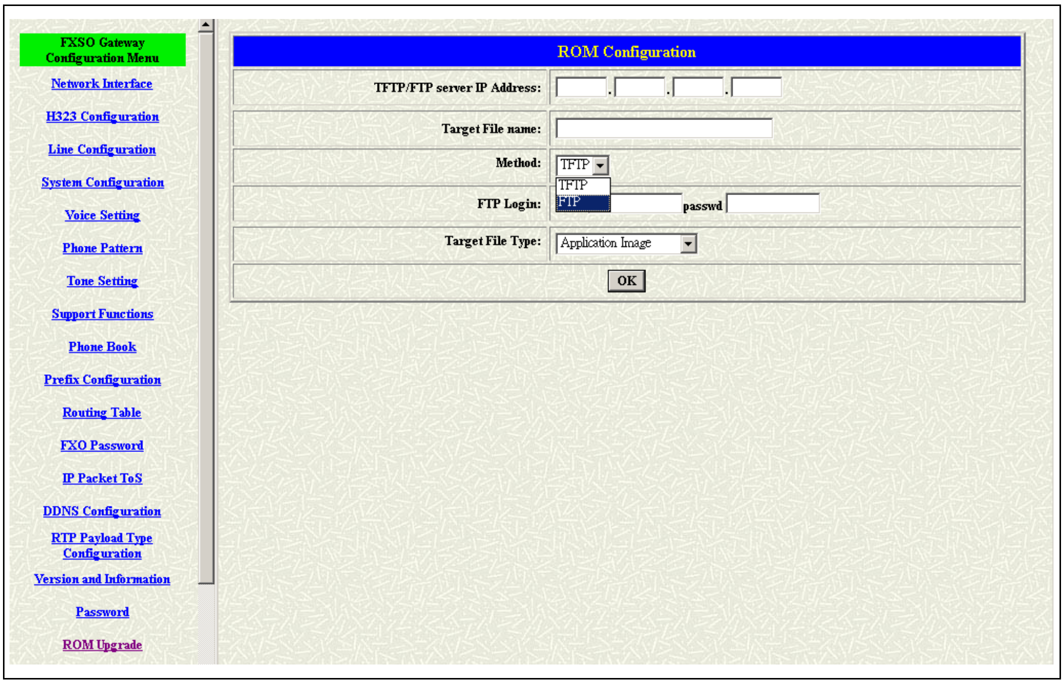


Figure 3.27: ROM Upgrade for FTP

2 Key in the IP address, the login name, password of your FTP server and the correct file name, file type. (see figure 3.28)

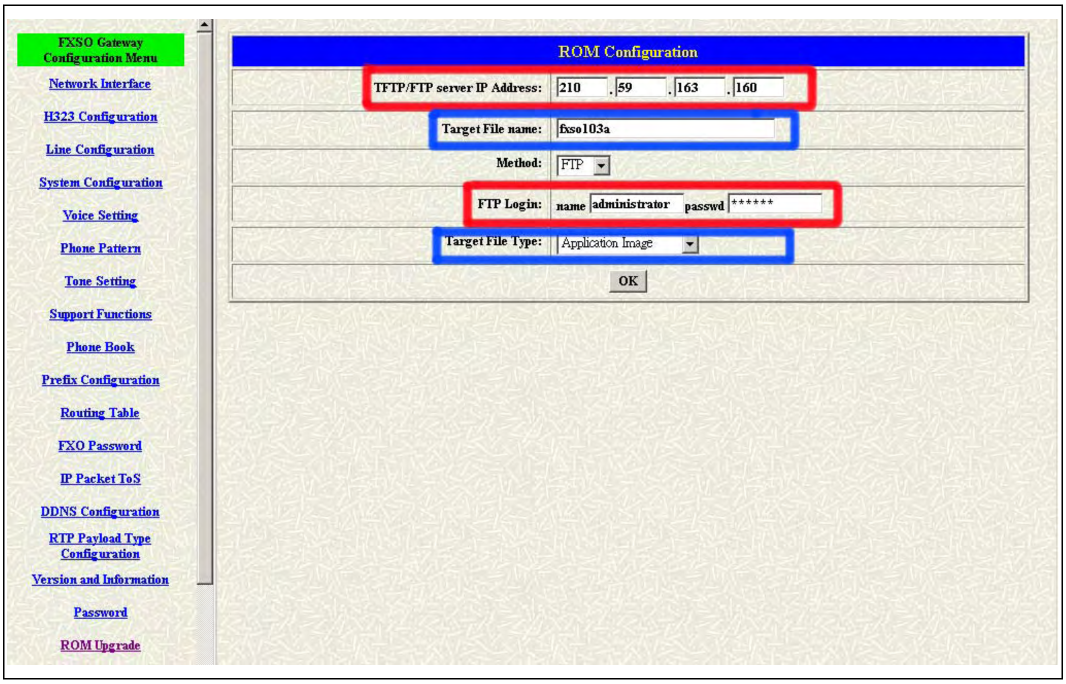


Figure 3.28: FTP information

3 Press the OK button to start the upgrade procedure.

□Updating the firmware by the TFTP server□

1 Downloading the TFTP program from our web site and install it first. Executing the TFTP program before you want to use the TFTP upgrade method.

2 Pick up the "Rom Upgrade" button to enter the upgrading web page and switch to the TFTP method. (see figure 3.29)

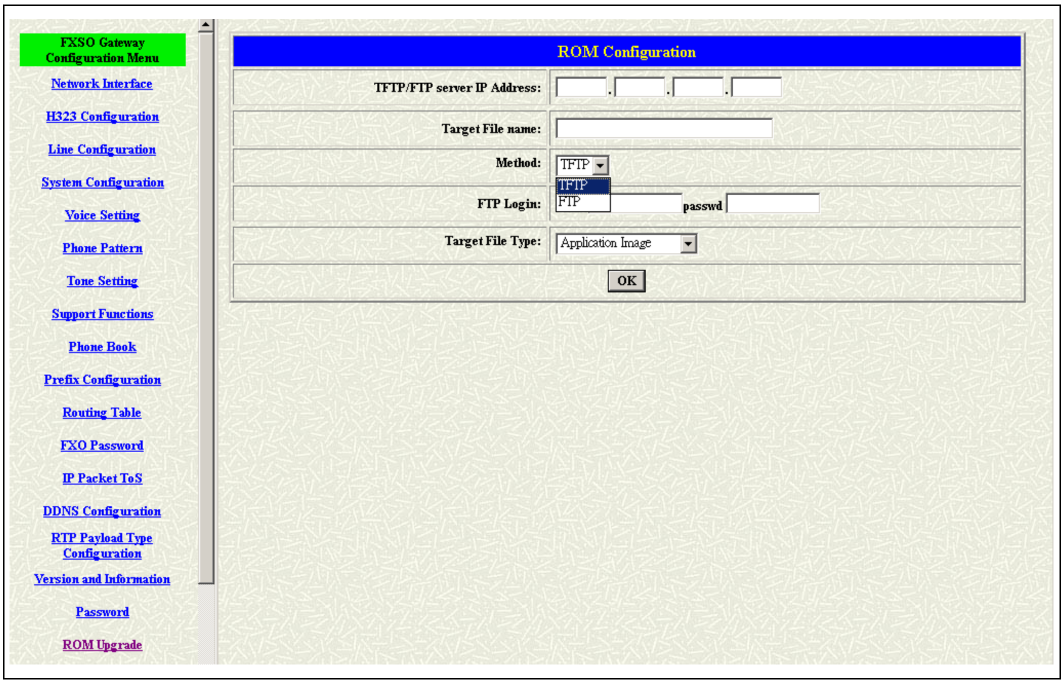


Figure 3.29: ROM Upgrade for TFTP

3 Key in the IP address of your TFTP server, pick up the file type for your upgrade file and the correct file name for upgrading. (see figure 3.30)

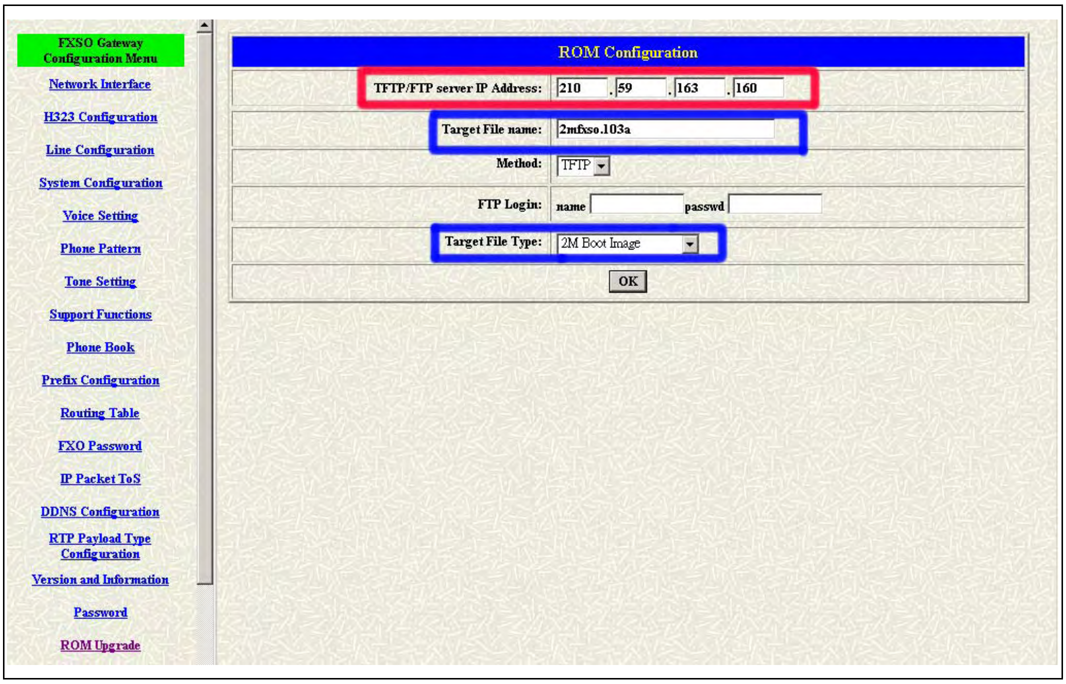


Figure 3.30: TFTP information

4 Press the OK button to start the upgrade procedure.

3.20 Flash Clean

Users could make all the configurations back to the default setting by this button. The password of the account and the networking configuration couldn't be back to the default setting by this command. (see figure 3.31)

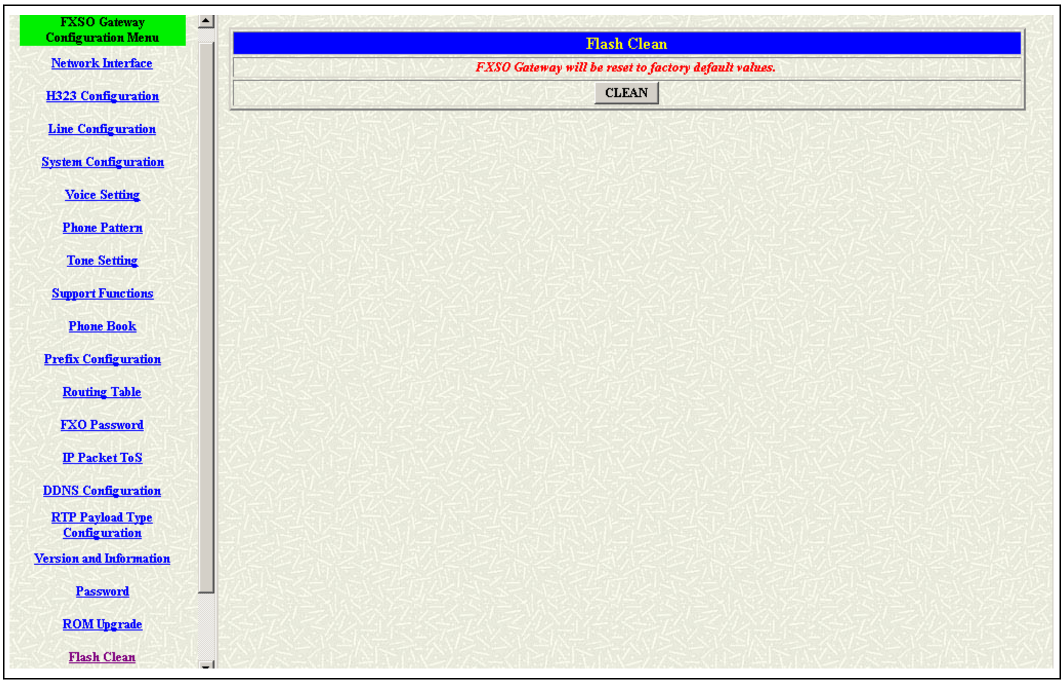


Figure 3.31: Flash Clean

3.21 Commit

This web page could save the configurations if users change some configurations. This is necessary for users change the configurations. (see figure 3.32)

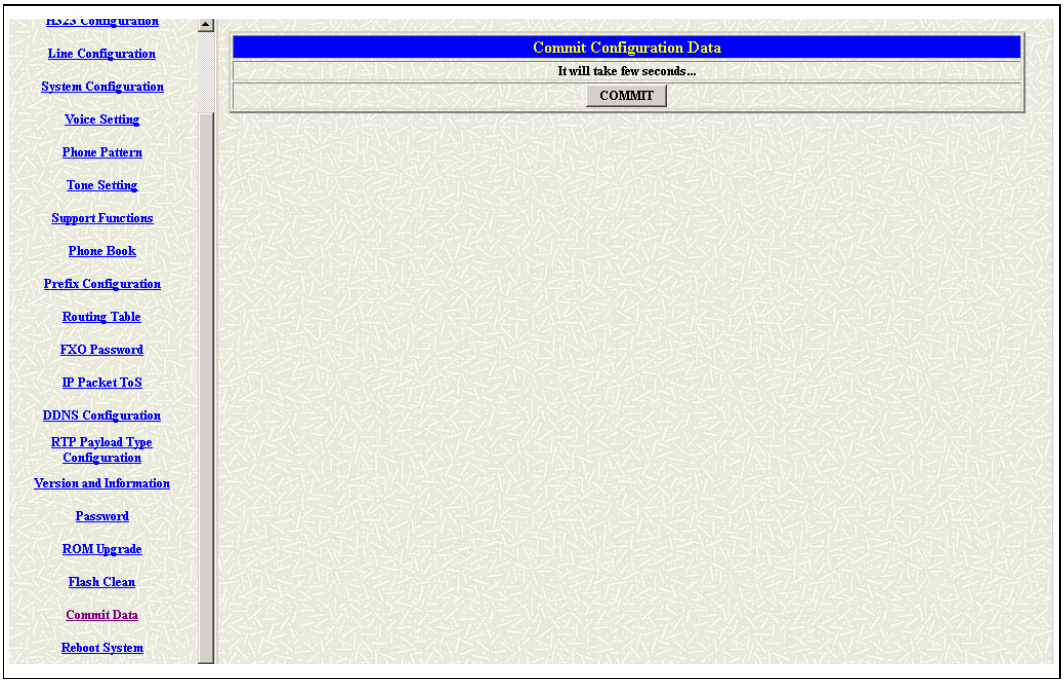


Figure 3.32: Commit

3.22 Reboot System

This web page will restart the whole system. This is the necessary step for the changing the configurations and makes it executed. (see figure 3.33)

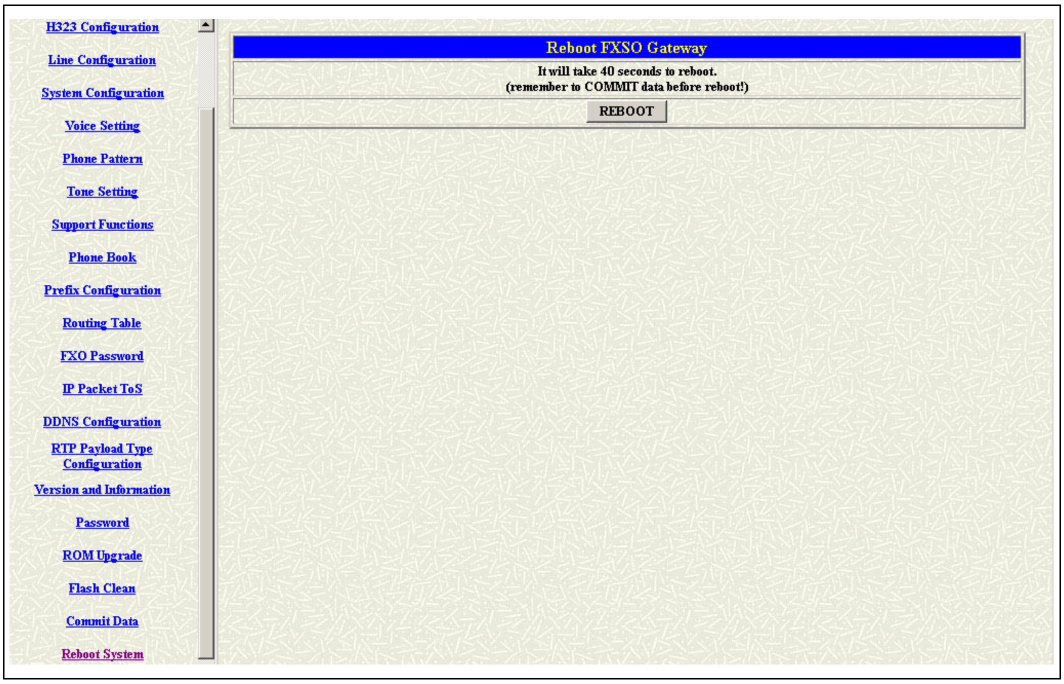


Figure 3.33: Reboot System

4. Command List

4.1 Hyper Terminal Setting

A terminal emulator is needed when using RS-232 port to configure Gateway. There are kinds of terminal emulator software. Here, we use Microsoft HyperTerminal to depict how to set up terminal emulator:

1 Execute the Hyper Terminal program, and then the following windows will pop-up on the screen. (START – Program files – Accessories – Communication – Hyper Terminal)

Figure 4.1: Hyper Terminal

2 Define a name such as ‘wg37’ for this new connection.

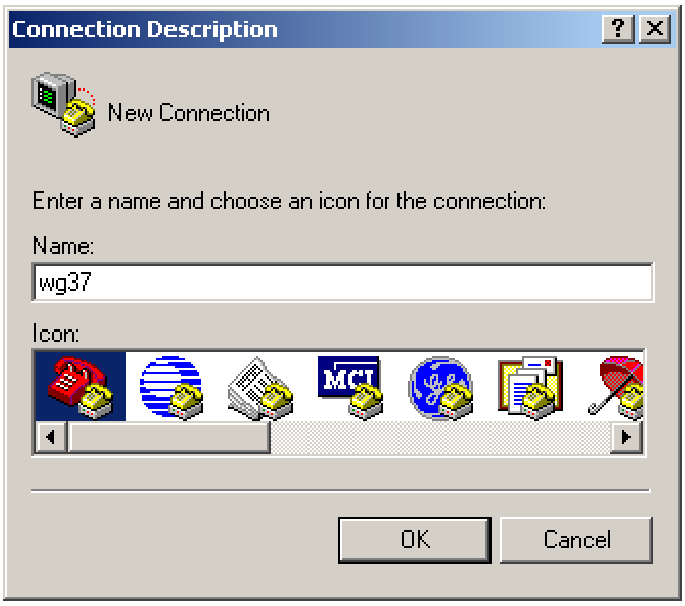


Figure 4.2: Edit the name of the connection

3 After pressing OK button, the next window appears, and then chooses ***COM1/2 Port***, which you are going to use.

Figure 4.3: Pick up the right interface to use

4 Configure the COM Port Properties as following:

- Bits per second: 9600
- Flow control: None

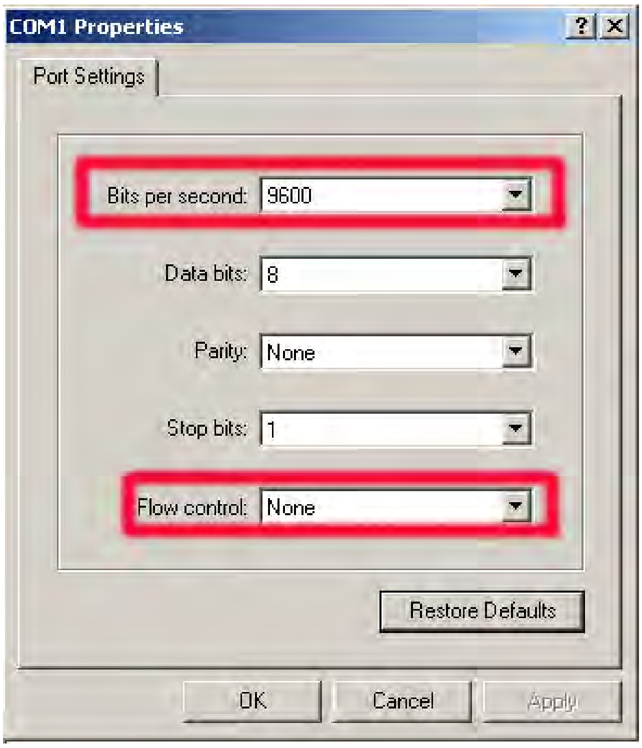


Figure 4.4: Configure the right Bps and control

5 Press ‘OK’ button, and then start to configure Gateway.

4.2 Command List

4.2.1 [help]

Type **help** or **man** or **?** to list all the available command.

```
usr/config$ help

help        help/man/? [command]
quit        quit/exit/close
debug       show debug message
reboot      reboot local machine
flash       clean configuration from flash rom
commit      commit flash rom data
ifaddr      Internet address manipulation
time        show current time
ping        test that a remote host is reachable
sysconf     System information manipulation
h323        H.323 information manipulation
line        Line information manipulation
route       Routing information manipulation
prefix      Prefix drop/insert information manipulation
pbook       Phone book information manipulation
voice       Voice information manipulation
support     Special Voice function support manipulation
phone       Setup of call progress tones and ringing
tone        Setup of disconnect tone
fxopwd      Setup of FXO password
record      Record voice for greeting and ask pin code
tos         IP Packet ToS (Type of Service) values
ddns        Dynamic DNS update manipulation
pt          DSP payload type configuration and information
rom         ROM file update
passwd      Password setting information and configuration

usage: help [command]
```

4.2.2 [quit]

Type **quit** will quit the Gateway configuration mode and turn back to login prompt (in console mode) or disconnect (in TELNET mode).

usr/config$ quit

Disconnecting...

login:

Note: It is recommended that type the "**quit**" command before you leave the console. If so, Gateway will ask password again when next user connects to console port.

4.2.3 [debug]

Open debug message will show up specific information while Gateway is in operation. After executing the debug command, it should execute command **debug -open** as well. One example is demonstrated below.

usr/config$ debug -add h323 vp h323vp

usr/config$ debug -open

In this example, user open debug flags including h323, vp, h323vp.

Parameters Usage:

-status Display the enabled debug flags.

-add Add debug flag.

-- h323 : h323 related information

-- vp : voice related information

-delete Remove specified debug flag.

-open Start to show debug messages.

-close Stop showing debug messages.

4.2.4 [reboot]

After **commit** command, type **reboot** to reload Gateway in new configuration. The procedure is as below:

usr/config$ reboot

.Attached TCP/IP interface to cpm unit 0
Attaching interface lo0...done

Hardware auto detect...
Hardware Type : 1FXS + 1FXO

HTTPD initialized...
cmInitialize succeed!
Ras port:1024
CallSignal port:1720

AC4804[0] is ok
successful 1 4
Initialize OSS libraries...OK!
VP v1.44 stack open sucessfully.

login::

4.2.5 [flash]

This command will clean the configuration stored in the flash ROM and reboot Gateway in factory default setting.

Parameter Usage:

-clean clean all the user defined values, and reboot Gateway in factory default mode.

Note: It is recommended that use "flash –clean" after application firmware id upgraded.

Warning: Only user who login with **root** can execute this command. Configurations of IP address and MAC address will be kept.

4.2.6 [commit]

Save changes after configuring Gateway.

----

usr/config$ commit

This may take a few seconds, please wait....
Commit to flash memory ok!
usr/config$

Note: Users shall use ***commit*** *to save modified value, or they will not be activated after system reboot.*

4.2.7 [ifaddr]

Configure and display Gateway network information.

---

usr/config$ ifaddr

LAN information and configuration
Usage:
ifaddr [-print]
ifaddr [-mode used]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway]
ifaddr [-sntp mode [server]]

-print	*Display LAN information and configuration.*
-mode	*Specify WAN ip mode.(0:Static / 1:DHCP /2:PPPoE)*
-ip	*Specify ip address.*
-mask	*Set Internet subnet mask.*
-gate	*Specify default gateway ip address*
-sntp	*Set SNTP server mode and specify IP address.*
-timezone	*Set local timezone.*
-ipsharing	*Specify usage of an IP sharing device and specify IP address.*
-ipchange	*Replace IP address if the shared IP is changed.*
-id	*PPPoE connection user name.*
-pwd	*PPPoE connection password.*
-reboot	*PPPoE connection password.*

Note:

Range of ip address setting (0.0.0.0 ~ 255.255.255.255).

SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).

Example:

ifaddr -ip 210.59.163.202 -mask 255.255.255.0 –gate 210.59.163.254

ifaddr -sntp 1 210.59.163.254

ifaddr -ipsharing 1 210.59.163.254

ifaddr –ipchange 1

usr/config$

Parameters Usage:

-print print out current [ifaddr] settings and status

-mode Define the network mode for this unit.

-ip assign IP address for Gateway

-mask assign internet subnet mask

-gate assign IP default gateway

-sntp Simple Network Time Protocol (1 = ON; 0 = OFF) When SNTP function is activated, users have to specify a SNTP server as network time source. An example is demonstrated below:

-timezone set local time zone according to GMT

-ipsharing Just put the static IP address for the WAN port of the IP sharing if the gateway is behind this IP sharing device.

----

usr/config$ ifaddr -ipsharing 1 210.11.22.33

If the IP address for the WAN port of the IP sharing device is dynamic. Please just enable this ipsharing function and leave the address with the empty. This dynamic is only support the GK from Asotel.

----

usr/config$ ifaddr -ipsharing 1

-ipchange   If the unit is behind the IP sharing device and the IP address for the WAN port of that IP sharing is using the dynamic IP address. This function has to be enabled.

----

usr/config$ ifaddr -ipchange 1

-id	This id is for the user name of the PPPoE usage.
-pwd	This password is for the user name of the PPPoE usage.
-reboot	If the connection disconnected by the ISP, the unit will reboot and get the ip again.

4.2.8 [time]

When SNTP function of Gateway is enabled and SNTP server can be found as well, type **time** command to show current network time.

----

usr/config$ time

Current time is THU JAN 01 05:29:23 1970

4.2.9 [ping]

Use **ping** to test whether a specific IP is reachable or not.

For example: if 192.168.1.2 is not existing while 192.168.1.254 exists. Users will have the following results:

---

usr/config$ ping 192.168.1.2

no answer from 192.168.1.2

usr/config$ ping 192.168.1.254

PING 192.168.1.254: 56 data bytes

64 bytes from 192.168.1.254: icmp_seq=0. time=5. ms

64 bytes from 192.168.1.254: icmp_seq=1. time=0. ms

64 bytes from 192.168.1.254: icmp_seq=2. time=0. ms

64 bytes from 192.168.1.254: icmp_seq=3. time=0. ms

----192.168.1.254 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)   min/avg/max = 0/1/5
usr/config$

4.2.10 [sysconf]

This command displays system information and configurations.

```
usr/config$ sysconf

System information and configuration
Usage:
  sysconf [-idtime digit][-keypad dtmf][-prefixsw digit]
          [-prefixdisab digit][-usrdefprefix digits][-eod digit]
          [-codec digit][-localrbt digit][-forwardtime digit]
          [-hwtype digit][-gwprefix digit][-ring on_time off_time]
          [-fxotype digit][-rba digit]
  sysconf -print

  -print          Display system overall information and configuration.
  -idtime         Inter-Digits time.(1~10 sec)
  -forwardtime    Forward time for FXS line if no answer.(5~65535 sec)
  -keypad         Select DTMF type: 0=In-band,
                                    1=H.245 Alphanumeric,
                                    2=H.245 SignalType,
                                    3=Q.931 UserInfo.
                                    4=RFC2833.
  -prefixsw       User defined local zone prefix switch.(ON:1 / OFF:0)
  -prefixdisab    Local zone prefix disable character.(one character
                  from 0~9, *, or NONE('-' key))
  -usrdefprefix   User defined local zone prefix.(0 ~ 20 digits)
  -codec          Codec select method.(Caller:0 / Master:1)
  -localrbt       Local ring back tone.(Enable:1 / Disable:0)
```

-gwprefix Drop gateway prefix when call from IP(Keep:0/Drop:1).
-hwtype Hardware type.(Auto:0 / 1FXS+1FXO:1 / 2FXS+2FXO:2)
-ring The ring time for ring detection.(Uint:ms)
-eod End of dial..(Enable:1 / Disable:0)
-fxotype FXO type.(Normal:0 / Force 2nd dial:1)
-rba Ring before answer.(For 1~10)

Example: sysconf -ring 500
usr/config$

Parameters Usage:

-print print out all current settings

-idtime set the duration(in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, it will dial out all number pressed. (1-10 seconds)

-forwardtime set forward time(5-65535 seconds)for FXS Line. If callee hasn't answered the call in this time, call will be forward to assigned number in [line] command. (please refer to ***[line]*** command for forward setting)

-keypad DTMF replay type. When value is "0", Gateway will transfer DTMF signal via In-Band type, "1" via H.245 UII Alphanumeric, "2" via H.245 UII Signal Type, and "3" via Q.931 UserInfo, and "4" via RFC2833 type. Users can adjust the value according to various applications.

-prefixsw switch on/off prefix function. If user enables prefix function, once user dials out, gateway will automatically add prefix number before number user dialed.

-prefixdisab set disable key (0,1,~9, *) to disable the prefix function in this current call. For example, if user has set prefix as 100, and wants to dial out 100123, user can only press 123, to dial out 100123.However, if user wants to dial 123 without prefix, user can press prefix disable key, for example "*", user can press "*", then dial 123, gateway will dial out 123 without adding prefix.

-usrdefprefix define prefix number.

Note: User can also define IP address here in P2P mode, once user press “#”, Gateway will call out this IP address.

Note:

“refixsw”, “prefixdisab”, “usrdefprefix” commands must work together.

-codec set who is the one to determine voice codec during negotiation. 0 is caller will determine the codec, and 1 is mater will determine the codec. (During negotiation 2 endpoints will compare gateway type level to determine who is master.)

-localrbt Enable/Disable locally generate ring back tone. “0” means gateway will receive ring back tone from remote callee, “1” means gateway will generate ring back tone locally.

-gwprefix drop or keep gateway prefix . “0” means when gateway has incoming call from IP side, it will not drop prefix before searching for callee number. “1” means when gateway has incoming call from IP side, it will drop prefix before searching for called number.

Note:

1. *If user wants to implement one-stage dialing in FXO Line, this function must be enabled, and user has to dial prefix number (instead of Line number of FXO Line)+ PSTN number to make a call to PSTN side connected with FXO Line.*
2. *After gateway-prefix-drop function is enabled, user must remember to re-configure line number of FXS Line, because line number of FXS Line must remove prefix number. For example, origin line number of FXS line is 1001, prefix is 100, since prefix number will be drop, once gateway has incoming call 1001, after drop gateway prefix 100, it will search line number “1”. So line number must be set as “1”.*

-hwtype application rom file of 37 series are the same no matter how many ports is the module, so after user downloads the application rom

file, user can select which hardware type is . ”0” means gateway will automatically detect the hardware type, “1” means the hardware type is 1FXS+1FXO, “2” means the hardware type is 2FXS+2FXO.

Note:

The default value is to auto detect hardware type. Usually it is not necessary to change this setting. Please make sure about your Hardware Type, Gateway may be not functional if set wrong hardware type.

-ring ring time for ring detection(in ms). When Gateway has incoming call from PSTN side to FXO port, Gateway will determine it is a ring but not noise only if it is longer than this ring time.

Note:

In Taiwan the ring time of PSTN usually is 1000ms, so if user set ring time longer that 1000ms, FXO port may not be able to pick up the call from PSTN side.

-eod It will transfer the DTMF in “#” if users disable the end of dial function. Users have to press the key pad in “#” if the end of dial function is enable.

-fxotype It’s for the 2nddial function. Any calls will be two-stage-dialing if user enable this function.

-rba Users could define the ring time if the calls coming from the PSTN side. This support 1 to 10 ring times.

4.2.11 [h323]

This command is for H.323 configuration related parameters.

----

usr/config$ h323

H.323 stack information and configuration

Usage:

-print Display H.323 stack information and configuration.

-mode Configure as Gatekeeper mode or Peer-to-Peer mode.

-gk Gatekeeper ip address. (0.0.0.0 ~ 255.255.255.255)

-algk	Second Gatekeeper ip address. (0.0.0.0 ~ 255.255.255.255)
-gwtype	Gateway Type (1 = Gateway and 0 = Terminal).
-dfgw	Default Gateway ip address (0.0.0.0 ~ 255.255.255.255).
-prefix	Prefix number.
-line1	Line 1(TEL 1) E.164 number.
-line2	Line 2(LINE 1) E.164 number.
-line3	Line 3(TEL 2) E.164 number.
-line4	Line 4(LINE 2) E.164 number.
-alias	IP side registered H323 ID.
-display	String representing display information for reporting to the called party.
-gkdis	Gatekeeper discovery (On=1, Off=0).
-gkid	Gatekeeper ID.
-rtp	RTP port number (1024~65532).
-ttl	RAS TTL time (0~3600 second).
-gkfind	Gatekeeper finding port (1024~65535).
-gkras	Gatekeeper RAS port (1024~65535).
-h225	H225 ras port (1024~65535).
-q931	H225 call signal port (1024~65535).
-dstq931	Destination H225 call signal port (1024~65535).
-range	Dynamically allocated port range (1024~65535).
-respto	Max waiting time for 1st response to a new call (1~200).
-connto	Max waiting time for call establishment after receiving 1st response of a new call (1~20000).
-passwd	H.235 security password.

Example:

h323 -gk 210.59.163.171 -line1 70 -line2 71 -line3 72 -line4 73

h323 -alias Your_Alias_Name -gkid GK -gkdis 1 -passwd 1234

usr/config$

Parameters Usage:

-print print current h323 related settings

-mode alternatives for gatekeeper or peer-to-peer mode (0=gatekeeper mode; 1=peer-to-peer mode). If users select gatekeeper mode, a valid gatekeeper is needed when Gateway is in operation.

----

usr/config$ h323 –mode 1 (peer to peer mode)

-gk to assign gatekeeper's IP address when Gateway is in gatekeeper mode.

-algk assign second gatekeeper's IP address as redundancy. If Gateway fails to register to main GK for 10 time, it will try to register to alternative GK.

-gwtype gateway type has two kinds, gateway and terminal. Gateway type – device will register as H.323 defined Gateway, user has to define [prefix] in next command.
Terminal type – device will register as H.323 defined Terminal, [prefix] command is not necessary.

-dfgw default gateway is applied under Peer-to-Peer mode. User defines a constant default gateway IP address, then any number dialed if the number is not listed in phone book table will pass forward to this IP Address.

-prefix assign Gateway prefix number, as well as the registered number on the Gatekeeper.

-line1 assign FXS TEL1 number.

-line2 assign FXO Line1 number.

-line3 assign FXS TEL2 number.

-line4 assign FXO Line2 number.

Note:

User can also set "x" in line number to disable the port. If the port is disabled, it can only receive calls but not calling out.

Note:

1. This is for Dynamix DW-0202/H, for Dynamix DW-0101/H model, there are only line1 and line2 command.
2. Line1~Line4 number must follow the prefix number if device is configured as **Gateway Type**. For example, if prefix number is 999, then the line1 & line2's number are 9991 & 9992.
3. If Gateway is configured as **Terminal Type**, each line will register to GK with it's own number, prefix number is not needed.
4. No matter in GK or P2P mode, user only needs to dial line number to reach local port. For example, in P2P mode, user wants to dial from FXS TEL1 to FXO Line1, only need to dial number of line2.

-alias	H.323 ID. If in gatekeeper mode, this h.323 ID must be different from others who are registering to the same gatekeeper.
-display	An addition name for special application if callee needs this r to recognize in called site.
-gkdis	set auto discovery function on or off. If this function is enabled and IP address of GateKeeper is set as 255.255.255.255, LAN Phone will multicast to search a GateKeeper on network with configured GateKeeper name (please refer to -gkid); if IP address of GateKeeper is set, before LAN Phone register to the assigned GateKeeper, it will send out GRQ(GateKeeper Request) message with configured GateKeeper name to GateKeeper first.
-gkid	set GateKeeper name for GateKeeper discovery. When Gateway send out GateKeeper discovery message will search GateKeeper with this GateKeeper name. (please refer to -gkdis)
-rtp	to allocate RTP port range—NOT RECOMMENDED. This may be used when RTP port range conflicts with Firewall policy. (each port of Gateway use 2 RTP ports)
-ttl	to set timer for TTL(Time To Live). Gateway would send RRQ, with keepAlive, to gatekeeper periodically according to TTL timer.
-gkfind	gatekeeper finding port. Port number, which Gateway uses it to

discover a gatekeeper. Default value is 1718.

-gkras to set default gatekeeper RAS port number. Default value, 1719, is well-known port for RAS communication.

-h225 to specify H.225 RAS port number.

-q931 to specify H.225 signal port number.

-dstq931 to specify destination H.225 signal port number.

-range To allocate port range (1500-65535) Gateway may use it.

-respto Maximum response time out

-connto Maximum connection time out

-passwd set H.235 security password. If user's GK need H.235 security token password to authenticate, user have to input topen password in this command. (please refer to your retailer of GK for information about token password if they implement this function in their GK)

Note: From –rtp to –conneto commands are for advanced users, please do not change the default settings if not necessary.

4.2.12 [line]

This command is for configure each line parameters of Gateway.

----

usr/config$ line

Gateway line information and configuration

Usage:

line -config number [hunt number][hotline number][forward number]

line -print Gateway line information.

hunt Hunting group.

hotline Hot line configuration.

forward No answer forward for FXS line.

Example:

line -config 1 hunt 1 hotline 1003 forward 1002

usr/config$

Parameter Usages:

-print print out all current settings of line

-config determine which line to configure

-hunt set hunting group flag of each line. User can assign different hunt group number represent different hunt group. For example, if user assigns FXS TEL1 as hunt group 1, and FXS TEL2 as hunt group 2, they will be determined as 2 different groups. On the other hand, if user assigns FXS TEL1 as hunt group 1, and FXS TEL2 as hunt group 1 too, when having incoming call to FXS TEL1, which is busy, this call will be route to FXS Line2.

Note: *FXO Lines and FXS TELs are treated as 2 different groups, so even they are in the same hunt group, call will only be routed to the same FXS or FXO Lines.*

-hotline set hotline table.The Hotline Mode is applied in limited two channels. User just picks up the phone set of one FXS TEL or calls in one FXO line, and gateway will automatically dial out a phone number. In the other hand, user will hear ring back tone or dial tone immediately depended on configurations of destination device.

Note: This function can both work in GK or P2P mode.

(1) Call out from FXS Line

GK Mode Usage:

Set gateway under gk mode.

Create a Hotline table with "***line*** " command.

----

usr/config$ line –config 1 hotline 1001

In this example means: if user picks up phone set of FXS Line1, gateway will automatically dial out "1001".

P2P Mode Usage:

Set gateway under P2P mode.

Create phone book table with "***pbook*** " command.

Create a Hotline table with "***line*** " command.

----

usr/config$ pbook –add name asotel ip 10.1.1.1 e164 1001
usr/config$ line –config 1 hotline 1001

In this example means: if user picks up phone set of FXS Line1, gateway will automatically dial out IP address of "1001".

(2) Call out from FXO Line

GK Mode Usage:

Set gateway under gk mode.
Create a Hotline table with "***line*** " command.

----

usr/config$ line –config 2 hotline 1001

In this example means: if user calls in FXO Line1, gateway will automatically dial out "1001".

P2P Mode Usage:

Set gateway under P2P mode.
Create phone book table with "***pbook*** " command.
Create a Hotline table with "***line*** " command.

----

usr/config$ pbook –add name asotel ip 10.1.1.1 e164 1001
usr/config$ line –config 2 hotline 1001

In this example means: if user calls in FXO Line1, gateway will automatically dial out IP address of "1001".

-forward set no answer forward table for FXS Lines.

Only **FXS** Lines provides No Answer Forward function. For call forward function, it can works under GK or P2P mode.

(1) GK Mode Usage:

----

usr/config$ line –config 1 forward 1002

In this example means: if user calls in FXS Line and hasn't been answered in forward time (please refer to ***[sysconf -forwardtime***] command), gateway will automatically forward this call to phone number "1002".

(2) P2P Mode Usage:

----

usr/config$ line –config 1 forward 1002

In this example means: if user calls in FXS Line1 when Line1 is busy, gateway will automatically forward this call to IP address of "1002" in Phone book.

4.2.13 [route]

This command is to set routing table for Gateway.

```
-------------------------------------------------------------------------------------------usr/
config$ route
Routing table information and configuration
Usage:
route -add [prefix number][dst number][e164 number]
        [min number][max number][hunt number]
route -delete index
route -modify index [prefix number][dst number][e164 number]
        [min number][max number][hunt number]
route -ip [dst number][e164 number]
route -fxs [dst number][e164 number]
route -fxo [dst number][e164 number]
route -print     Routing table information.
        prefix    The prefix of dialed number.
        dst        Destination port(FXS:0/FXO:1/IP:2).
        e164       Destination e164 number(when destination is FXS or
```

FXO).

min *Min digits.(0 ~ 255)*

max *Max digits.(0 ~ 255)*

hunt *Hunt method for busy forward(NONE:0 / GROUP:1 / ALL:2)*

Example:

route -add prefix 100 dst 0 e164 1001 min 1 max 3 hunt 1

route -ip dst 0 e164 1001

route -fxs dst 2

route -fxo dst 0 e164 x

route -modify 1 prefix 100 dst 0 e164 1001 min 1 max 3 hunt 1

route -delete 1

usr/config$

Parameter Usages:

-print print out all routing table information

-add add a routing rule in routing table. User can add less than 50 rules. (***route –add prefix "prefix number" dst "destination port type" e164 "e.164 number of port" min "minimum digits needed" max "maximum digits can't be exceeded"***)

-delete delete a routing rule in routing table (***route –delete "index of routing rule"***)

-modify modify a routing rule in routing table. (***route –modify "index of routing rule" prefix "prefix number" dst "destination port type" e164 "e.164 number of port" min "minimum digits needed" max "maximum digits can't be exceeded"***)

-ip create routing table for incoming call from IP side. (***route –ip dst "destination port type" e164 "e.164 number of port"***)

-fxs create routing table for incoming call from FXS TELs. (***route –fxs dst "destination port type" e164 "e.164 number of port"***)

-fxo create routing table for incoming call from FXO Lines. (***route –fxo dst "destination port type" e164 "e.164 number of port"***)

prefix prefix of dialed number

dst destination port, 0 means FXS TELs, 1 means FXO Lines, 2

means IP side, x means no determinate number.

e164 destination e.164 number. This only need to be set when routed port is FXS TELs or FXO Lines to determine which port will this call be routed to.

min minimum digits needed.

max maximum digits needed.

hunt set hunt method for busy forward. 0 means no hunting, 1 means hunting method follows the rule of ***[line]***, 2 means hunting method is to hunt between all ports in the same type, for example, destination port is FXS TEL will hunt in all FXS TELs, destination port is FXO Lines will hunt in all FXO Lines.

Usage Example:

1. ***route –add prefix 100 dst 0 e164 1001 min 1 max 3 hunt 1***
   This command means if gateway has incoming call's prefix number is 100, and total digits is between 1 to 3, this call will be routed to FXS TEL 1001, and if TEL 1001 is busy, call will be routed to another FXS TEL.
2. ***route –ip dst 1 e164 1002***
   This command means incoming call from IP side will be routed to FXO Line of number 1002.
3. ***route –fxs dst 1 e164 1002***
   This command means incoming call from FXS TELs will be routed to FXO Line of number 1002.
4. ***route –fxo dst 2***
   This command means incoming call from FXO Lines will be routed to IP side.

Note:

(1) When destination is IP side, e.164 number doesn't need to determine. (Ex. route –fxs dst 2)

(2) If user doesn't want to determine a specific port to route, e.164 number must set as "x". (Ex. route –ip dst 1 e164 x)

(3) Default value: Incoming call from FXS and FXO ports will be

forward to IP side.

4.2.14 [prefix]

This command is for make rules for drop or insert prefix digits.

----

usr/config$ prefix

Prefix drop/insert information and configuration

Usage:

prefix -add [prefix number][drop number][insert digits]

prefix -delete index

prefix -modify index [prefix number][drop number][insert number]

prefix -print　　Prefix drop/insert information.

prefix　　The prefix of dialed number.

drop　　Drop prefix(Enable:1/Disable:0).

insert　　Insert digits.

Example:

prefix -add prefix 100 drop 1 insert 2000

prefix -add prefix 100 drop 1

prefix -add prefix 100 drop 0 insert 200

prefix -delete 1

prefix -modify 1 prefix 100 drop 0 insert 300

usr/config$

Parameter Usages:

-add　add a rule to drop or insert prefix digits of incoming call.(***prefix –add prefix "prefix number" drop 0/1 insert "insert number"***)

-delete　delete a rule to drop or insert prefix digits of incoming call. (***prefix –delete prefix "prefix number"***)

-modify　modify a rule to drop or insert prefix digits of incoming call. (***prefix –modify prefix "prefix number" drop 0/1 insert "insert number"***)

prefix　set which prefix number to implement prefix rule.

drop　enable or disable drop function.If this function is enabled, Gateway will

drop prefix number on incoming call.

insert set which digit to insert on incoming call.

4.2.15 [pbook]

Phone Book function allows users to define their own numbers, which mapping to real IP address. It is effective only in peer-to-peer mode. When adding a record to Phone Book, users also have to reboot the machine, and the record will be effective immediately.

```
-----------------------------------------------------------------------------------------
usr/config$ pbook

Phone book information and configuration
Usage:
 pbook     [-add [name string][e164 number][ip address]
           [port number][drop digit][insert number]]
           [-modify number [name string][e164 number][ip address]
           [port number][drop digit][insert number]]
           [-delete number]
 pbook   -print

 -print          Display phone book information and configuration.
 -add             Add new phone book record)
 -delete         Delete phone book record
 -modify         Modify phone book record.
                   name    : 1 ~ 10 characters.
                   e164    : 1 ~ 10 digits.
                   ip        : IP adress.
                   port     : 1024 ~ 65535.
                   drop    : 0:Disable/1:Enable.
                   insert : 1 ~ 10 digits.
Example:
  pbook -add name test e164 1234 ip 192.168.1.10 port 1720 drop 1 insert
              5678
```

pbook -delete 1

pbook -modify 1 name test e164 5678 ip 192.168.1.10 port 1720 drop 0

usr/config$

Parameter Usages:

-print print out current contents of Phone Book. (***pbook -print***) Users can also add *index number*, from 1 to 100, to the parameter to show specific phone number. (Ex. ***pbook –print 1***)

Note: <index number> means the sequence number in phone book. If users do request a specific index number in phone book, Gateway will give each record a automatic sequence number as index.

-add add a new record to phone book. When adding a record, users have to specify ***name***, ***ip***, and ***e164*** number to complete the command.

name name to represent callee.

e164 e.164 number for mapping with IP address of callee

ip ip address of callee

port call signal port number of callee

drop drop e.164 number when dial out. 0 means to keep e.164 number, 1 means to drop e.164 number when dialing out.

insert insert digits.(1~10 digits)

-delete delete a specific record. "pbook –delete 3" means delete **index 3** record.

-modify modify an existing record. When using this command, users have to specify the record's index number, and then make the change.

PhoneBook Rules:

The e164 number defined in phone book will fully carry to destination. It is not just a representative number for destination's IP Address. In other words, user dial this e164 number to reach destination, destination will

receive the number and find out if it is matched to its e164, including Line number in some particular device.

4.2.16 [voice]

The voice command is associated with the audio setting information. There are four voice codecs supported by Gateway.

```
usr/config$ voice

Voice codec setting information and configuration
Usage:
voice [-send [G723 ms] [G711A ms] [G711U ms] [G729 ms] [G729A ms]
[G729B ms] [G729AB ms] ]
        [-volume [voice level] [input level] [dtmf   level]]
        [-nscng [G711U used1] [G711A used2] [G723 used3]]
        [-echo used] [-mindelay t1] [-maxdelay t2] [-optfactor f]
voice -print
voice -priority [G723] [G711A] [G711U] [G729] [G729A] [G729B] [G729AB]

    -print      Display voice codec information and configuration.
    -send         Specify sending packet size.
                  G.723   (30/60 ms)
                  G.711A (20/40/60 ms)
                  G.711U (20/40/60 ms)
                  G.729   (20/40/60 ms)
                  G.729A (20/40/60 ms)
                  G.729B   (20/40/60 ms)
                  G.729AB (20/40/60 ms)
    -priority   Priority preference of installed codecs.
                  G.723
                  G.711A
                  G.711U
                  G.729
                  G.729A
```

G.729B
G.729AB

-volume *Specify the following levels:*
voice volume (0~63, default: 28),
input gain (0~63, default: 28),
dtmf volume (0~31, default: 23),

-nscng *No sound compression and CNG. (G.723.1 only, On=1, Off=0).*

-echo *Setting of echo canceller. (On=1, Off=0, per port basis).*

-mindelay *Setting of jitter buffer min delay. (0~150, default: 90).*

-maxdelay *Setting of jitter buffer max delay. (0~150, default: 150).*

Example:

voice -send g723 60 g711a 60 g711u 60 g729 60 g729a 60 g729b 60 g729ab 60
voice -volume voice 20 input 32 dtmf 27
voice -echo 1 1
usr/config$

Parameters Usage:

-print print current voice information and configurations.

-send define packet size for each codec. 20/40/60ms means to send a voice packet per 20/40/60 milliseconds. The smaller the packet size, the shorter the delay time. If network is in good condition, smaller sending packet size is recommended. In this parameter, 20/40/60ms is applicable to G.711u/a law, and G.729/G.729A/G.729B/G.729AB codec, while 30/60ms is applicable to G.723.1 codec.

-priority codec priority while negotiating with other h323 device. This parameter determines the listed sequence in h.245 TCS message. The codec listed in left side has the highest priority when both parties determining final codec. User can also select the particular codec without others.

----

usr/config$ voice –priority g723 (only select this codec)

---

usr/config$ voice –priority g723 g729 g711u g711a (select four codecs, and g723 is the first choice)

Note:

(1) For Dynamix DW-0202/H there are 2 version of Application rom, please check version of Application rom (rom –print). If the version is 2fxso729.102, Dynamix DW-0202/H doesn't have codec G.723.1. If the version is sfxso723.102, Dynamix DW-0202/H doesn't have codec G.729 series.

(2) For Dynamix DW-0101/H, the Application rom has the only one version which is named fxso.102 provide all codec.

-volume There are three adjustable value. **voice volume** stands for volume, which can be heard from Gateway side; **input gain** stands for volume, which the opposite party hears; **dtmf** volume stands for DTMF volume/level, which sends to its own Line.

Note, level of volume is too high or too low may be result in bad performance while connecting to each other.

-nscng silence suppression and comfort noise generation setting (1 = ON; 0 = OFF). It is applicable to G.723 codec only. An example is demonstrated below:

----

usr/config$ voice -nscng g723 1

-echo activate each canceler (1 = ON; 0 = OFF).

-mindelay the minimum jitter buffer size. (Default value= 90 ms)

-maxdelay the minimum jitter buffer size. (Default value= 150 ms)

----

usr/config$ voice –mindelay 90 –maxdelay 150

Note: be sure to know well the application before you change ***voice*** *parameters because this might cause incompatibility.*

4.2.17 [support]

This command provides some extra functions that might be needed by users.

```
usr/config$ support

Special Voice function support manipulation
Usage:
support [-t38 enable][-t38ecm enable][-faxrdd digits]
          [-fstart enable][-tunnel enable][-h245fs enable]
support -print
      -t38          T.38(FAX) enabled/disabled.
      -t38ecm       T.38(FAX) ECM enabled/disabled.
      -t38asn1      T.38(FAX) ASN.1 support enable/disable.
      -faxrdd       FAX redundancy depth(0 ~ 2).
      -fstart       Fast start enabled/disabled.
      -tunnel       H245 Tunneling enabled/disabled.
      -h245fs       H245 message after FastStart support enabled/disabled.
      -earlyh245 EarlyH245 support enabled/disabled.
Example:
      support   -t38 1
      support   -t38ecm 1
      support   -t38asn1 1
      support   -faxrdd 1
      support   -fstart 1
      support   -tunnel 1
      support   -h245fs 1
      support   -earlyh245 1

usr/config$
```

Parameter Usages:

-print print current settings in **support** command.

-t38 to switch ON/OFF (1 = ON; 0 = OFF) T.38 fax ability. When T.38 ability is ON, Gateway will automatically defer codec (G.723 or G.729 series) to T.38 when FAX signal is detected.

-t38ecm to switch ON/OFF (1 = ON; 0 = OFF) T.38 fax Error Correction Mode ability. When high-speed T.38 FAX is running, Gateway will automatically execute ECM function.

-t38asn1 Enable the ASN.1 support with the FAX.

-faxrdd set fax redundancy depth. User can increase FAX redundancy depth when network traffic is heavy. For example, if user set fax redundancy as 2, Gateway will resend fax packets every 2 packets.

-fstart to switch ON/OFF (1 = ON; 0 = OFF) fstartStart function. Fast Start function can shorten the connection time if the opposite party also supports fastStart.

-tunnel to switch ON/OFF (1 = ON; 0 = OFF) H.245 tunneling function. If the function is ON, Gateway will send H.245 (Call Control messages) via H.225's (Call Signal messages) link. The function is effective only when both sides support H.245 tunnel.

-h245fs If the function is ON, Gateway can send H.245 messages after FastStart.

-earlyh245 enable/disable early H.245 function. The function is effective only when both sides support early H.245.

Note: it is not recommended to change the value in this command, only if users do know well the application. This might cause incompatibility with other devices.

4.2.18 [phone]

Gateway's progress tone is configurable. Default tone value is set according to U.S. tone specification. Users may adjust the values according to their own country's tone specification or users-defined tone specification.

```
usr/config$ phone

Phone ringing , ringback tone , busy tone , dial tone setting and notes
Usage:

phone [-ring [freq     ][ringON    ][ringOFF ][ringLevel]]
        [-rbt   [freqHi ][freqLo    ][freqHiLev][freqLoLev]
                 [Tone1ON][Tone1OFF][Tone2ON    ][Tone2OFF ]]
        [-bt     [freqHi ][freqLo    ][freqHiLev][freqLoLev]
                 [Tone1ON][Tone1OFF][Tone2ON    ][Tone2OFF ]]
        [-dt     [freqHi ][freqLo    ][freqHiLev][freqLoLev]
                 [Tone1ON][Tone1OFF][Tone2ON    ][Tone2OFF ]]
        [-flash [freqLo ][freqHi ]]
phone [-print [ring]\[rbt]\[bt]\[dt]\[flash]]

       -print    Display phone ringing/tone configuration.
                  ring :     ringing
                  rbt   :     ringback tone
                  bt     :     busy tone
                  dt     :     dial tone
                  flash:     flash tone
        -ring     ringing configuration set .
        -rbt       ringback tone configuration set .
        -bt         busy tone configuration set .
        -dt         dial tone configuration set .
        -flash    flash configuration set .
```

Note:

ringing frequency		: 15 ~ 100	(Unit : Hz)
ringing ring ON/OFF		: 0 ~ 8000	(Unit : ms)
ringing level		: 0 ~ 94	(Unit : V)
tone	frequency	: 0 ~ 65535	(Unit : Hz)
tone	freqLevel	: 0 ~ 65535	(Unit : mVrms)

---

tone Tone ON/OFF : 0 ~ 8000 (Unit : ms)

Example:

phone -print rbt
phone -ring 20 2000 4000 94
phone -rbt 480 440 8 8 2000 4000 2000 4000
phone -bt 620 480 8 8 500 500 500 500
phone -dt 440 350 8 8 500 1023 1023 1023
phone -flash 100 300

usr/config$

Parameters Usage:

-print print current call progress tone configurations (**ring** – ring tone, **rbt** – ring back tone, **bt** – busy tone, **dt** – dial tone, flash – flash). This parameter should be accompanied with tone type. For example:

----

usr/config$ phone -print rbt

Phone ring back tone paramter

Ringback Tone frequency high : 480
Ringback Tone frequency low : 440
Ringback Tone frequency high level : 13
Ringback Tone frequency low level : 13
Ringback Tone tone1 on : 100
Ringback Tone tone1 off : 200
Ringback Tone tone2 on : 1023
Ringback Tone tone2 off : 1023

usr/config$

Note:
For tone simulation, Gateway adopts dual frequencies as traditional telephone does. If users want to have their own call progress tone, they can change the value of tones. High and Low frequency/level/cadence can be configured respectively.

-ring to set RING tone value.
The played tone type, when Gateway is receiving a call.

-rbt to set RingBackTone value
The played tone type, when Gateway receives a Q.931 Alerting message. In condition that Gateway is the originate side.

-bt to set BusyTone value.
The played tone type, when destination is busy.

-dt to set DialTone value.
The played tone type, when hook off a phone set of workable Gateway.

-flash set the detective flash range in ms, for example, 300-500 ms.

4.2.19 [tone]

This command is basically for FXO ports.

```
usr/config$ tone

Disconnect tone and remote ring back tone configuration
Usage:

tone   [num][freqHi ][freqLo   ][freqHiLev][freqLoLev]
              [Tone1ON][Tone1OFF][Tone2ON   ][Tone2OFF ]]
tone   -print   Display disconnect tone configuration.
       [num]    Tone index(1~4:Disconnect tone / 5~8:Remote ring back
tone).
Example:
       tone -print
       tone 1 620 480 8 8 50 50 1023 1023

usr/config$
```

Parameter Usages:

-print show all tone configuration

[num] tone index. 1~4 is disconnect tone, 5~8 is remote ring back tone.

For FXO ports Gateway must detect disconnect tone to determine when to disconnect the call, so user must set disconnect tone of PBX or PSTN network connected to FXO ports.

When making a call from FXO ports, there are 2 ways to detect callee has already picked up the call, one is to detect reverse signal, the other is to detect the termination of ring back tone, so user must set ring back tone of PBX or PSTN network.

(If user doesn't know about the frequency of disconnect tone or ring back tone, please refer to ***[record]*** command to detect frequency.)

For each tone may has 1 set or 2 sets (high and low) of frequencies. If user wants to set 0 in on/off time, please set "1023" represent "0". (ex. ***tone 1 620 480 8 8 50 50 1023 1023***)

(***tone "index of tone" "frequency of high" "frequency of low" "level of high" "level of low" "on time of high" "off time of high" "on time of low" "off time of low"***)

4.2.20 [fxopwd]

This command is for FXO ports.

----

usr/config$ fxopwd

FXO password information and configuration

Usage:

fxopwd -add [passwd number][direction number]

fxopwd -delete index

fxopwd -modify index [passwd number][direction number]

fxopwd -print FXO password information.

passwd The password.

direction   FXO direction(Out:0/In:1/Both:2).

Example:

fxopwd -add passwd 1234 direction 0

fxopwd -delete 1

fxopwd -modify 1 passwd 1234 direction 2

usr/config$

Parameter Usages:

-print   show all FXO password configuration

-add   add 1 set of FXO password

-delete   delete 1 specific set of FXO password

-modify   modify 1 specific set of FXO password

passwd   password

direction   the direction number which password implement. 0 means the calls from IP side to PSTN side, 1 means the calls from PSTN side to IP side, 2 means both sides. Once user enables FXO password function, caller will be asked to enter password when call in PSTN side or IP side according to the direction of password being implemented. (***fxopwd –a passwd "password" direction 0/1/2***)

Note:

User can set total 10 sets of password, no matter what the direction is.

4.2.21 [record]

User can record greeting and askpin file and analyze tone frequency by calling in FXO line of Gateway.

----

usr/config$ record

Recoed greeting voice and ask pin code voice, tone analize.

Usage:

record -greeting filename

-askpin filename

-tone

Example:

record -greeting greeting.100

record -askpin askpin.100

record -tone

usr/config$

Parameter Usages:

-greeting record greeting file. User must assign a file name for greeting, once record is finished, file recorded will be display in rom –print.

----

usr/config$ record -greeting test.100

Please off hook TEL 1 and press (N) for next step...

n

Press (R) to start record...

r

Press (S) to stop record...

...

...

...s.........

...

....

Press (P) to play the voice or (W) to write to flash or (Q) to quit...

p

w

Please wait a moment...

Write flash ok...

Boot Rom : sdboot.200
Application Rom : fxso.100
DSP App : 48302ce3.300
DSP Kernel : 48302ck.300
DSP Test Code : 483cbit.bin
Greetings : test.100
Ask Pin : askpin.100
q
usr/config$

-askpin record askpin file. User must assign a file name for askpin file, once record is finished, file recorded will be display in rom –print.

----

usr/config$ record -askpin askpintest

Please off hook TEL 1 and press (N) for next step...
n

Press (R) to start record...
r

Press (S) to stop record...
..
..
..
..
..
.........................s...
..

...

.

Press (P) to play the voice or (W) to write to flash or (Q) to quit...

p

w

Please wait a moment...

Write flash ok...

Boot Rom : sdboot.200

Application Rom : fxso.100

DSP App : 48302ce3.300

DSP Kernel : 48302ck.300

DSP Test Code : 483cbit.bin

Greetings : greeting.100

Ask Pin : askpintest

q

usr/config$

Note: Remember to press enter after press any command.

-tone analyze tone frequency. Gateway can analyze tone frequency as user provide tone in FXO Line1.

----

usr/config$ record -tone

Press (R) to start record...

r

...

...

...

....

Analyzing!! Please wait a moment……..
Frequency 1 : 480
Frequency 2 : 620
0.25sec on 0.25sec off
usr/config$

Note:

1. ***Record ring back tone: user can use FXS Line1 to call FXO Line1, after hearing ring back tone, use this command to detect frequency of ring back tone.***
2. ***Record disconnect tone: Please read the procedure of recording disconnect tone file from the web site in application.***
3. ***The value of disconnect tone and ring back tone will not write in flash automatically. Please use the command in "tone" to write in the tone table.***

The Procedures of recording the disconnect tone

Before you start

A PSTN line which connect with the Line 1 port.
A analog phone connect with the Tel 1 port.
Configure Peer-to-Peer mode.

Please record the disconnect tone just follow the stage as below

1. Please enter the command before you record the disconnect tone
   record –tone
2. Make a call from PSTN side into Line 1 port.
3. You will get a greeting when the call enter the gateway.
4. Pease dial the number of the Tel 1 port.
5. The phone will ring if the number you dial is correct.

6. Pick up the phone and make sure the call is connect.
7. Hang up the phone which is from PSTN side and Tel 1 port will get the disconnect tone.
8. When you get the disconnect tone from the phone set of the Tel 1 port, press <**R**> and <**ENTER**> buttons to start recording the disconnect tone.
9. Please hang up the phone if you get the message as below□*Analizing!! Please wait a moment…*
10. There are three values you will get after analyzing. Please leave the value which is over 1000 Hz, this is not the frequency of disconnect tone.
11. Please put the frequency in the tone table just follow the command□ *tone 4 420 680 8 8 25 25 50 50*

Example-1

(Make a call from PSTN to FXO port)
usr/config$ record -tone

Press (R) to start record...
(Please make sure that you are already finish the steps 2 ~ 7)
r (Press “Enter” button after you key in “R”)
....
....
....
....

Analizing!! Please wait a moment...
(You coule hang up the call from PSTN if you get this message)

Frequency 1 : 481
Frequency 3 : 621
0.25sec on 0.25sec off

tone 4 481 621 8 8 25 25 1023 1023

(Put this value in to the tone table)

tone –print

Disconnect tone 1 paramter
    Frequency high : 620
    frequency low : 480
    frequency high level : 8
    frequency low level : 8
    Tone1 on : 25
    Tone1 off : 25
    Tone2 on : 1023
    Tone2 off : 1023
Disconnect tone 2 paramter
    Frequency high : 450
    frequency low : 0
    frequency high level : 8
    frequency low level : 0
    Tone1 on : 35
    Tone1 off : 35
    Tone2 on : 1023
    Tone2 off : 1023
Disconnect tone 3 paramter
    Frequency high : 620
    frequency low : 480
    frequency high level : 8
    frequency low level : 8
    Tone1 on : 50
    Tone1 off : 50
    Tone2 on : 1023
    Tone2 off : 1023
Disconnect tone 4 paramter
    Frequency high : 621

frequency low : 481
frequency high level : 8
frequency low level : 8
Tone1 on : 25
Tone1 off : 25
Tone2 on : 50
Tone2 off : 50
(Confirm the values is correct or not)

(Key in the commit and reboot command if you finish the procedures as above)

□Example-2□

(Make a call into FXO port)
usr/config$ record -tone

Press (R) to start record...
(Please make sure that you are already finish the steps 2 ~ 7)
r (Press "Enter" button after you key in "R")
....
....
....
....

Analizing!! Please wait a moment...
(You could hang up the call from PSTN if you get this message)

Frequency 1 : 473
Frequency 2 (2623) is more than 1000, please ignore it.
0. 25sec on 0.25sec off

tone 4 473 473 8 8 25 25 1023 1023
(Please configure the high and low frequency as the same value if you just get

a singal frequency)

tone –print

Disconnect tone 1 paramter

Frequency high	: 620
frequency low	: 480
frequency high level	: 8
frequency low level	: 8
Tone1 on	: 25
Tone1 off	: 25
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 2 paramter

Frequency high	: 450
frequency low	: 0
frequency high level	: 8
frequency low level	: 0
Tone1 on	: 35
Tone1 off	: 35
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 3 paramter

Frequency high	: 620
frequency low	: 480
frequency high level	: 8
frequency low level	: 8
Tone1 on	: 50
Tone1 off	: 50
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 4 paramter

Frequency high	: 473
frequency low	: 473

frequency high level : 8
frequency low level : 8
Tone1 on : 25
Tone1 off : 25
Tone2 on : 50
Tone2 off : 50

(Confirm the values is correct or not)

(Key in the commit and reboot command if you finish the procedures as above)

4.2.22 [tos]

IP Packet ToS(type of Service)/Differentiated Service configuration.

----

usr/configtos

IP Packet ToS(type of Service)/Differentiated Service configuration
Usage:
tos [-rtptype dscp]
tos [-sigtype dscp]
tos -print
[-rtpreliab mode]
tos -print

Example:
tos -rtptype 7 -sigtype 0

Parameter Usages:
-rtptype the packages of voice
-sigtype the package of call signal

Note:
The value of rtptype and sigtype is from 0 to 63.
It's working if it supported by your network.

4.2.23 [ddns]

The dynamic DNS service information and configuration

----

usr/config$ ddns

The dynamic DNS service information and configuration

Usage:

ddns [-print]

ddns [-enable 0/1]

ddns [-serve Address] [-hostname Name] [-id ID]

[-passwd Password]

[-checkip option] [-checkipsrv Address]

[-delay time]

[-force IP]

-print	*Display Dynamic DNS information and configuration.*
-enable	*1:Enabled/0:Disable the dynamic DNS service.*
-server	*Specify DDNS server address.*
-hostname	*Registered domain name.*
-id	*Registered account ID.*
-passwd	*Registered account password.*
-checkip	*1:Enabled/0:Disable check the host current IP address.*
-checkipsrv1	*Specify IP address check server.*
-checkipsrv2	*Specify secondary IP address check server.*
-delay	*Setting the service delay time.(1~59 minutes or 1~24 horus)*
-force	*Force execute the dynamic DNS service.*

Example:

ddns -print

ddns -enable 1

ddns -server member.dyndns.org -hostname ipphone.dyndns.org

ddns -delay 30 m (30 minutes)

ddns -delay 12 h (12 hours)

ddns -force 11.22.33.44

Parameter Usages:

-enable enable the DDNS function

-server enter the server address of thr DDNS server you use

-hostname enter the domain name address which you get from the DDNS server

-id key in your id

-passwd key in your password

-checkip enable or disable the check current user's ip address

-checkipsrv1 enter the ip address of the check server

-checkipsrv2 the secondary ip address of the check server

-delay service delay time

-force execute the DDNS function all the times

4.2.24 [pt]

RTP payload type configuration and information

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usr/config$ pt

RTP payload type configuration and information

Usage:

pt -print Display the RTP payload type information

-rfc2833 Configure the DTMF RFC2833 payload type

-dtmf Configute the DTMF payload type

-fax Configure the FAX payload type

-faxbypass Configure the FAX ByPass payload type

-modembypass Configure the MODEM ByPass payload type

-redundancy Configure the Redundancy payload type

-modemrelay Configure the MODEM Relay payload type

Example:

pt -rfc2833 96 -fax 101

usr/config$

4.2.25 [rom]

ROM file information and firmware upgrade function.

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usr/config$ rom

ROM files updating commands
Usage:
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp][-greet][-askpin]
      -s TFTP/FTP server ip -f filename
rom -print
      -print       show versions of rom files. (optional)
      -app         update main application code(optional)
      -boot        update main boot code(optional)
      -boot2m      update 2M code(optional)
      -dsptest     update DSP testing code(optional)
      -dspcore     update DSP kernel code(optional)
      -dspapp      update DSP application code(optional)
      -greeting    update greeting voice file(optional)
      -askpin      update ask pin code voice file(optional)
      -s           IP address of TFTP/FTP server (mandatory)
      -f           file name(mandatory)
      -method      download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
      -ftp         specify username and password for FTP
Note:
      This command can run select one option in 'app', 'boot',
      , 'dsptest', 'dspcore', and 'dspapp'.
Example:
      rom -method 1
      rom -ftp vwusr vwusr
      rom -app -s 192.168.4.101 -f app.bin

usr/config$
```

Parameter Usages:

-print show versions of all rom files

-app, boot, boot2m, dsptest, dspcore, dspapp, greeting, askpin to update main Application program code, Boot code, DSP testing code, DSP kernel code, DSP application code, greeting file, askpin file.

-s to specify TFTP server's IP address when upgrading ROM files.

-f to specify the target file name, which will replace the old one.

-method to decide using TFTP or FTP as file transfer server. "0" stands for TFTP, while "1" stands for FTP.

-ftp if users choose FTP in above item, it is necessary to specify pre-defined username and password when upgrading files.

4.2.26 [passwd]

For security concern, users have to input the password before entering configuration mode. "**passwd**" command is for password setting purpose.

```
usr/config$ passwd

Password setting information and configuration
 Usage:
   passwd -set Loginname Password
   passwd -clean
 Note:
   1. Loginname can be only 'root' or 'administrator'
   2. passwd -clean will clear all passwd stored in flash,
      please use it with care.
 Example:
   passwd -set root Your_Passwd_Setting

usr/config$
```

Parameter Usages:

-set

(***passwd –set "login name" "password"***)

Note : "login name" can be "***root***" or "***administrator***" only. "root" and

“administrator” have the same authorization, except some commands that can be executed by “root” only – “***passwd –clean***”, “***rom –boot***”, ”***rom –bot2m***” and “***flash –clean***”.